SOUND DEVICES

664

Portable Production Mixer and Recorder
User Guide and Technical Information
firmware rev. 1.04

Sound Devices, LLC
E7556 State Rd. 23 and 33 • Reedsburg, WI • USA
+1 (608) 524-0625 • fax: +1 (608) 524-0655
Toll-Free: (800) 505-0625
www.sounddevices.com
support@sounddevices.com

CF UDMA SD TIME CODE

Table of Contents

Front Panel Descriptions 1
Left Panel Connectors and Control 2
Right Panel Connectors and Controls 3
Rear Panel Descriptions 4
Screen Overview 5
Main Screen 5
RTN Screen 5
Input Setup and Control 6
Input Settings Screen 6
Phantom Power 8
Gain/Trim and Fader Relationship 8
Input Trim (Analog)8
Input Trim (Digital)8
Input Fader8
High-Pass Filter 9
Pan Control 9
Input Linking 9
M/S Matrixing10
Digital Inputs 10
Limiters 11
Metering 12
Meter Ballistics 12
VU12
Peak + VU12
Peak Only12
Peak Hold12
Input Activity LED 13
Headphone Peak LED 13
Headphone Monitoring 13
Headphone Gain 13
Headphone Source Selection 13
Headphone Presets14
Headphone Tones 14
PFL (Channel Solo Monitor) 15
RTN and COMM Monitoring 15
Output Setup and Control 16
Master and Aux Outputs 16
XLR16
TA3 (Master L, R, X1, and X2)16
Hirose 10-pin16
Direct Outputs16
Tape Output16
Master and Aux Bus Level 17
AES Digital Outputs 17
COM Setup 17
COM Send Program 18
Com Program Auto-Mute 18
Slate/Com Mic & Tone Oscillator 18
Slate Microphone 18
Internal Slate Mic19
External Slate Mic19
Tone Oscillator 19
Alternate Slate/Com Switch 19
Digital Audio Recorder 20
Transport Control 20
Recording Tracks 20
Track Arming21
Track-to-Media Routing21
Track Naming22
Sampling Rate 22
Word Clock Synchronization23
Bit Depth 23
Bit Depth and Dynamic Range23
Recording Media 23
Time Code 24
Timecode modes24
Timecode Frame Rate25
Jam Menu26
Timecode Hold Off26
F Sampling Rate Modes 26
Pre-Roll 27
Synchronization 27
File Management 28
File List28
File Type29
File Structure29
Folder Options30
Automatic File Splitting30
Metadata Fields30
Sound Reports31
Take Management 32
Take List32
Take Number33
Scene Name33
Take Notes and Phrases34
Playback 35
Using a USB Keyboard 35
Mixer Linking 36
Linking the 664 With Another 664 or 552 Mixer 36
Linking to Other Mixers 36
Linking to a 302 or 44236
Linking to a MixPre-D or Original MixPre37
Powering 37
External Powering 37
Internal Battery Powering 37
Voltage Metering 38
Power Consumption 38
Storing and Recalling Settings 39
Setup Menu 40
Inputs 40
Outputs 40
Limiters 41
Recorder 41
Comms/Returns 42
Timecode/Sync 42
File Storage 43
System 44
Quick Setup 45
Front Panel Button Shortcuts 46
Connector Pin Assignments 47
Routing Diagram 49
Specifications 50
Analog Inputs 50
Analog Outputs 50
Digital Outputs/Recorder 51
Timecode and Sync 51
Power 52
Environmental 52
Dimensions and Weight 52
Accessories 53
Wave Agent 54
Declaration of Conformity 55
Warranty and Technical Support 56

Copyright Notice and Release

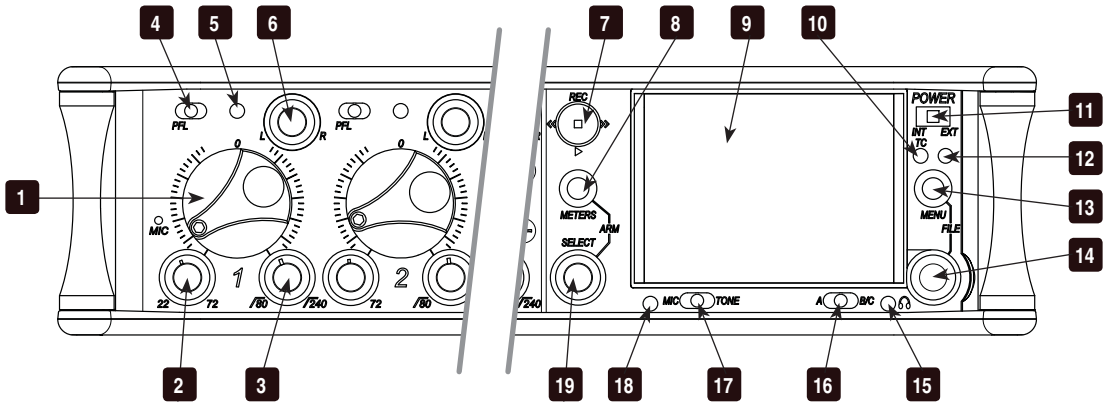
All rights reserved. No part of this publication may be reproduced, stored in a retrieval system, or transmitted in any form or by any means, electronic, mechanical, photocopying, recording, or otherwise, without the expressed written permission of SOUND DEVICES, LLC. SOUND DEVICES is not responsible for any use of this information.

SOUND DEVICES, LLC shall not be liable to the purchaser of this product or third parties for damages, losses, costs, or expenses incurred by purchaser or third parties as a result of: accident, misuse, or abuse of this product or unauthorized modifications, repairs, or alterations to this product, or failure to strictly comply with SOUND DEVICES, LLC's operating and installation instructions.

Microsoft Windows is a registered trademark of Microsoft Corporation. Macintosh is a registered trademark of Apple Computer. Other product and company names mentioned herein may be the trademarks of their respective owners.

The sound waves logo is a registered trademark of Sound Devices, LLC.

Front Panel Descriptions



1) **Input Fader**
Primary control for adjusting the level of an input during operation. Ranges from off to +15 dB. Nominal setting is in the middle (0 dB).

2) **Gain (Trim)**
Coarse input gain control. Sets the initial input sensitivity level so that the Input Fader can be used for fine gain adjustments. Range is from +22 dB to +72 dB. *See Input Setup and Control.*

3) **Highpass Filter Control**
Adjusts corner (-3 dB) frequency of high-pass filter. Full counter-clockwise position (detented) deactivates the High-Pass Filter. Range is 80-240 Hz, 12 dB/oct to 6 dB/oct. *See Input Setup and Control.*

4) **PFL / Input Select Switch**
Slide left: Pre-Fade Listen. Sends the input's pre-fade signal to HP monitor mono mix. The 664 supports simultaneous PFL of multiple inputs. Does not affect Master Output signal. Slide the switch left to activate, and again to deactivate. For momentary action, hold the switch left for one second or longer. The Input LED flashes yellow when an input's PFL is active. **Slide right:** Input Settings. Enters the Input Settings Screen where basic input setup and input-to-output bus routing is performed. *See Input Setup and Control.*

5) **Input LED**
Indicates input signal activity. Illuminates in various colors and intensities to show signal level and activity. Green = signal presence (pre-fader), yellow = limiter activity (pre- and post-fade), red = signal overload/clipping (pre- and post-fade), flashing yellow = input PFL.

6) **Input Pan**
Controls the Left/Right balance of the input signal to the Stereo Master Bus.

7) **Transport Control**
Controls the Integrated Digital Recorder. Slide up to Record, press in to Pause/Stop, slide down to Play, slide left to Rewind, slide right to Fast Forward. *See Digital Audio Recorder.*

8) **Meters Button**
Displays the Main Screen which includes metering, filename, time code and other important information. Toggles to RTN Screen when pressed from Main Screen. Returns to Main Screen from any other Screen.

9) **LCD**
Displays contextual operating information and user interface.

10) **Internal Timecode LED**
When the 664 is powered down and timecode Mode is Freerun or 24-Hour Run, the Internal Timecode LED will flash blue to indicate that timecode is being maintained. The 664 will hold accurate timecode for 2 hours after being powered down.

11) **Power Switch**
Three-position slide switch, selects between AA battery power or external DC sources, middle position is off.

12) **Power LED**
Illuminates green to indicate the 664 is powered on.

13) **Menu Button**
Displays the Setup Menu.

14) **Headphone Encoder**
Meter and Input Settings Screen: Turn to adjust headphone gain. In MaiPush to select headphone monitor source. In Menus turn to navigate. Push to make selection. In Input Settings Screen push to select input source.

15) **Headphone Clipping LED**
Illuminates red to indicate headphone output is approaching clipping level.

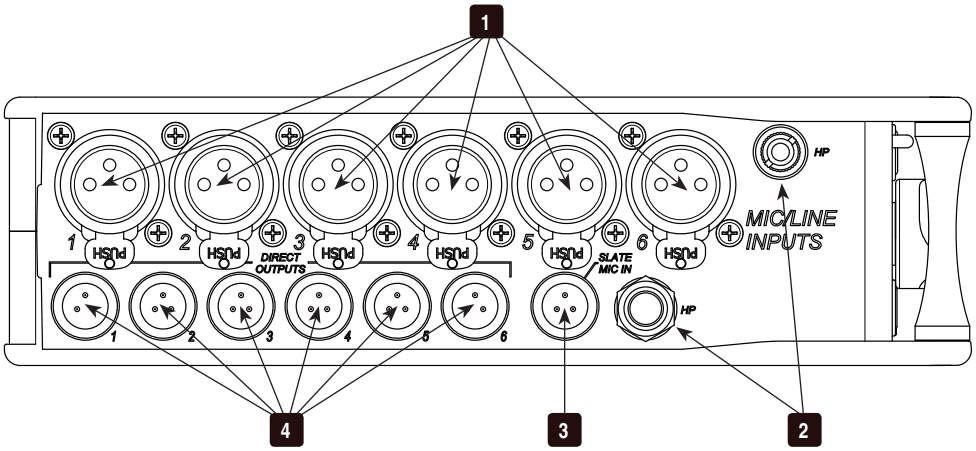
16) **RTN Switch**
Slide Left to activate RTN A, slide Right to activate RTN B. To access secondary function Press and hold Select Encoder then slide left for COM RTN and right for RTN C. Primary and Secondary functions of the RTN toggle switch can be selected in the Setup Menu.

17) **Slate / Tone Switch**
Slide left to activate the Slate Microphone, slide again to deactivate. For momentary action hold for one second or longer. Slide right to activate the Tone Oscillator. Tone will latch if held for one second or longer; slide again to deactivate. Secondary function: press and hold Select Encoder then slide Mic switch to activate COM. Primary and Secondary functions can be switched in the Setup Menu.

18) **Slate Mic LED**
Illuminates green when Slate Mic or COM is active.

19) **Select Encoder**
Multi-function encoder. Selects Tracks and RTN's on the Main and RTN Screen. On the Main Screen, press the Select Encoder and the Meters button to arm/disarm record tracks. With L, R, X1, X2, or any RTN selected, press then turn to adjust level. Vertical Scroll in matrix windows. Turn to adjust digital trim level in Input Settings Window. Turn to scroll cursor and press to insert a space character during text entry. Acts as shift button to access secondary functions.

Left Panel Connectors and Control



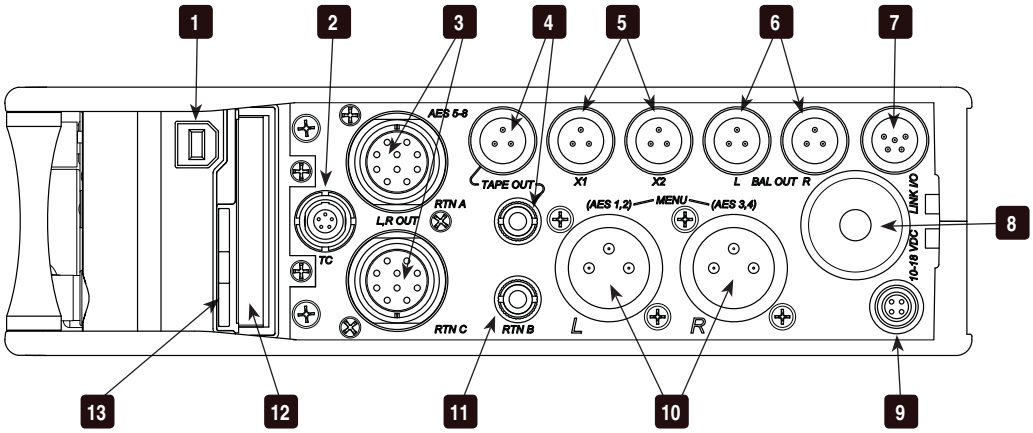
1) **XLR Analog Inputs 1-6**
Active-balanced analog microphone- or line-level input for inputs 1-6 on XLR connector. Input type is set within the Input Settings Screen. Inputs 1 and 6 can also accept AES3 or AES42 signal. Pin-1 ground, pin-2 (+), pin-3 (-).

2) **Headphone Output**
1/4-inch and 3.5 mm TRS stereo headphone connectors. Can drive headphones from 8 to 1000 ohm impedances to very high levels. Tip = left, ring = right, sleeve = ground.

3) **Slate Mic Input**
TA3 input for connecting external slate microphone. Select between internal or external slate mic (with or without 12V phantom) from the Setup Menu section **COMMS/RETURNS**. Pin-1 ground, pin-2 (+), pin-3 (-).

4) **TA3 Direct Outputs 1-6 / Inputs 7-12**
Balanced direct outputs on TA3 connectors. Direct output signal is pre- or post-fader and level is selected between Line, -10, and Mic levels in the Setup Menu section **OUTPUTS**. Tone signal appears at the direct outputs. With the CL-6 Input Expander attached, these connections can be selected as analog line inputs 7-12. Pin 1 = Ground; pin 2 = Hot (+); pin 3 = Cold (-) float pin 3 to unbalance.

Right Panel Connectors and Controls



1) **USB B Connector**
Factory use and keyboard connection (with adapter).

2) **Timecode I/O**
Time code input and output on 5-pin LEMO® connector.

3) **10-pin A and C**
Each connection includes a pair of transformer-isolated Outputs and a stereo unbalanced Return input. Analog Output levels are selected between Line, -10, and Mic levels in Setup Menu section **OUTPUTS**. 10-pin A outputs can be set to AES Outputs 5,6 and 7,8 in the Setup Menu section **OUTPUTS**.

4) **Tape Output**
Unbalanced stereo, tape level output on TA3 (Pin 1 = Ground, pin 2 = Left, pin 3 = Right) and 3.5 mm (Sleeve = Ground, Tip = Left, Ring = Right) connector.

5) **X1 and X2 Outputs**
Line, -10, or Mic level selected in the Setup Menu section **OUTPUTS**. Pin 1 = Ground, pin 2 = Hot (+), pin 3 = Cold (-) float pin 3 to unbalance.

6) **TA3 Master Outputs**
Line, -10, or Mic level selected in the Setup Menu section **OUTPUTS**. Pin 1 = Ground, pin 2 = Hot (+), pin 3 = Cold (-) float pin 3 to unbalance.

7) **Link I/O**
Used to link additional Sound Devices 664, 552, 302, 442, or MixPre mixers.

8) **Battery Compartment**
Holds five AA (LR6) batteries for backup powering. NiMH rechargeable cells advised.

9) **DC Input**
Accepts DC voltages from 10–18 V for powering. Pin 1 = Negative (–), pin 4 = Positive (+). Ext DC is fully isolated (floating) from the rest of the circuitry.

10) **XLR Master Outputs**
Transformer-balanced analog outputs on standard 3-pin XLR connectors. Pin 1 = Ground; pin 2 = Hot (+); pin 3 = Cold (-). Unbalance by grounding pin 3 to pin 1. Can be set to send AES3 digital signals (1,2 and 3,4 on L and R respectively) in Setup Menu section **OUTPUTS**.

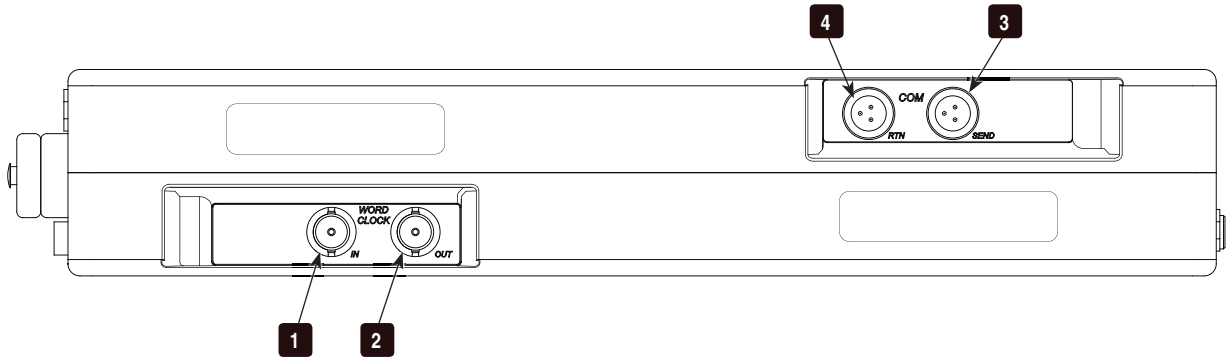
11) **RTN B Input**
Unbalanced stereo 3.5 mm female connector for Return B audio input. Sleeve = Ground, Tip = Left, Ring = Right.

12) **CompactFlash Slot**
Accepts approved CompactFlash cards with the label-side toward the rear of the 664. Compatible with Type I and Type II cards. High-speed UDMA cards are recommended for higher track count recording.

13) **SD Card Slot**
Accepts SD/SDHC/SDXC cards with the notched corner oriented toward the top of the 664. High speed class 10 cards are recommended. Insert until it clicks securely in the slot. The card should glide smoothly into the slot. Push to eject.

Visit www.sounddevices.com/approved for an up-to-date list of tested and approved cards.

Rear Panel Descriptions



1) **BNC Word Clock Input**
Accepts word clock rates between 32 kHz and 48.048 kHz for synchronizing the internal recorder to external digital audio devices.

2) **BNC Word Clock Output**
Provides word clock signal to synchronize external digital audio devices to the 664.

3) **TA3 COM Send**
Unbalanced, stereo, line-level output. Program assigned to COM output from Setup Menu.

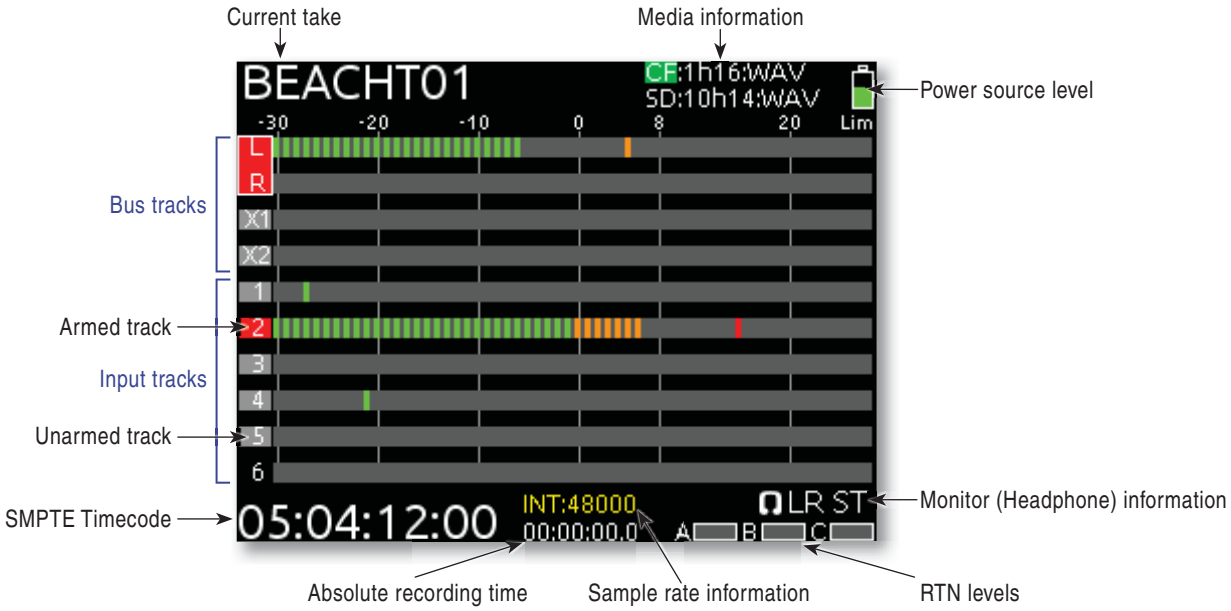
4) **TA3 COM Return**
Line-level input for return feed from on-set communications sources.

Screen Overview

Main Screen

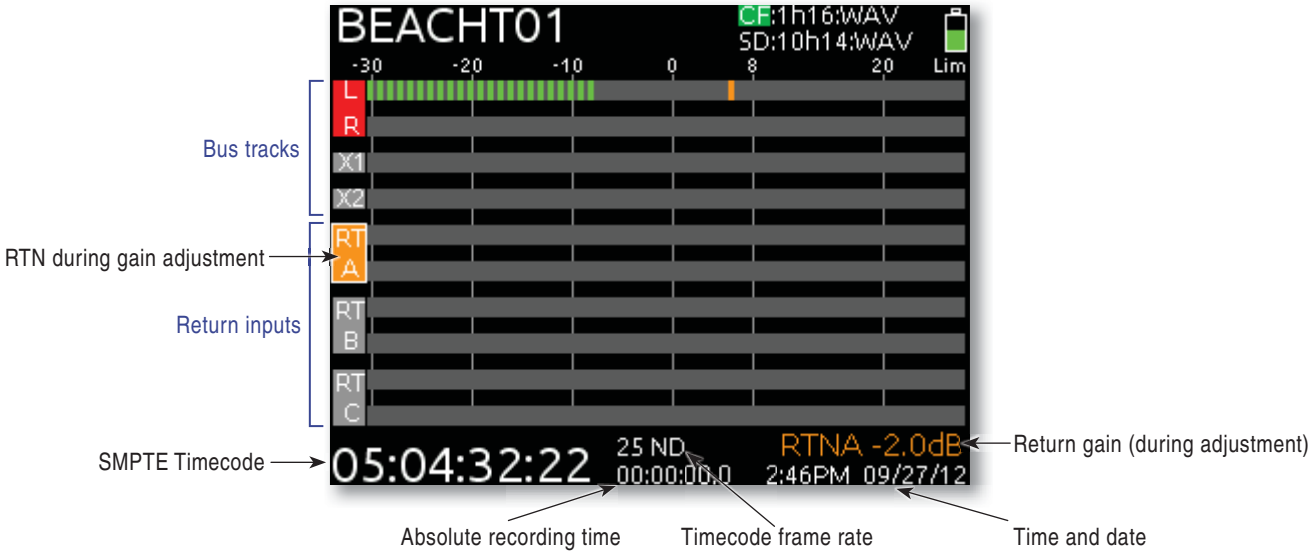
The Main Screen displays important information at-a-glance. The Main Screen can be accessed quickly from any other screen by pressing the Meters button.

Display of the sample rate information and timecode frame rate information can be "swapped" by holding the Meters button and Sliding the RTN Switch to the left.



RTN Screen

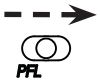
When the Meters button is pressed from the Main Screen, the RTN Screen is shown.



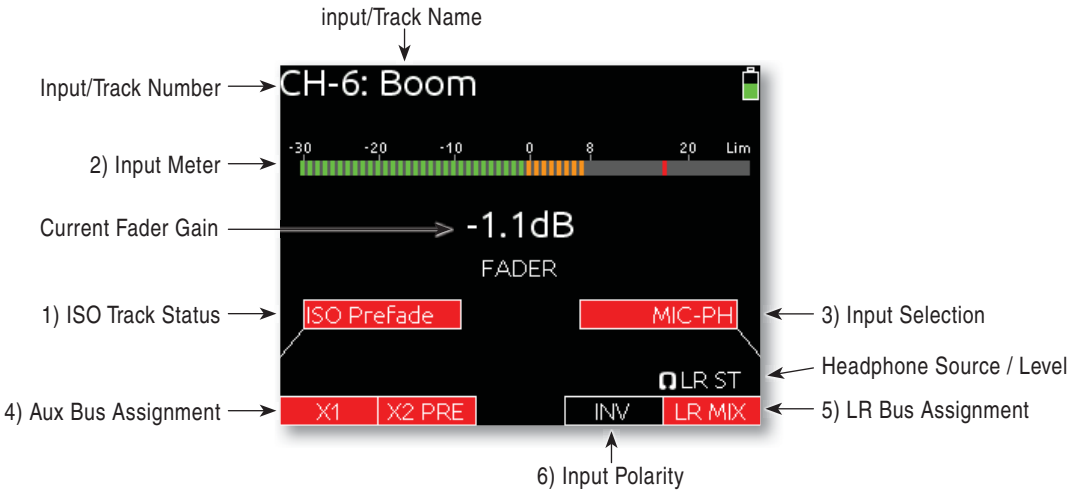
Input Setup and Control

The 664 has six, full-featured audio inputs on XLR connectors (six additional inputs on TA3 connectors with the optional CL-6 Input Expander attached). Each analog input has a wide gain range to accommodate nearly all signal types, from microphones to line-level sources. Inputs can be used as either balanced or unbalanced connections. To unbalance, tie pin-3 to pin-1 of the XLR connector. There is no change in gain between unbalanced and balanced connections into the 664.

Input Settings Screen



Each input's settings are accessed through its Input Settings Screen. To access the Input Settings screen for an input, slide the Input Selector Switch to the right. The LCD will display information pertaining to the selected input and provide access to.



1) ISO Track Status

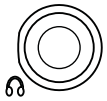
Displays the status of the Input's dedicated Iso Track on the recorder. Each input is permanently routed to its Iso track. To select whether an input is routed pre- or post-fade, push the Select Encoder, turn it to make a selection, and push it again to confirm the selection.

The background color of the Iso Track Status box indicates the status of the Input's Iso Track. A blue background indicates that the Iso Track is not active and will not be recorded. A red background indicates the Iso Track is armed for recording.

2) Input Meter

Displays the Input's signal level and limiting activity. The level displayed is the level to the Iso Track and will be pre- or post-fade depending on the Iso Track Status. Meter ballistics can be set globally from VU or Peak+VU in Setup Menu item **SYSTEM > Meter Ballistics**.

3) Input Selection



Displays the available input types. To change the input type, push the Headphone Encoder, turn it to make a selection, and push it again to confirm the selection.

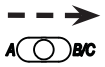
Input Type	Description
Off	Input off.
MIC	Use with dynamic microphones or other mic-level signals.
MIC-PH	Use with phantom-powered condenser microphones only. Provides 48V or 12V phantom power. *see Phantom Power*
LINE	For use with any line-level source.
LINE-PH	For use with phantom-powered condenser microphones only. Provides 48V or 12V phantom power, but at a line-level gain range. Useful for recording extremely loud sounds.
AES42	Digital input with power activated for digital microphones, inputs 1, 2, 5, and 6 only.
AES3	Digital input, inputs 1, 2, 5, and 6 only

AES42 and AES3 option only appears in input 2 when selected in input 1, and only appears in input 5 when selected in input 6. Channel 1 and Channel 6 XLR connectors are used for digital inputs.

4) Aux Bus Assignment

Displays the status of the Input's assignment to the X1 and X2 Output Buses. Inputs 2 through 5 can only be routed to X1 and X2 post-fade, while Inputs 1 and 6 can be routed to X1 and X2 pre- or post-fade. Slide the Slate / Tone Switch left for the X1 Track and right for the X2 Track to cycle through the available options.

5) LR Bus Assignment



Displays the status of the Input's assignment to the main Left and Right Output Buses. Inputs routed to the Left and Right Output Bus are always post-fader and post-pan. To add or remove the Input from the Left and Right Output Bus, slide the RTN Switch to the right.

6) Input Polarity (Inputs 2, 4, and 6)

Polarity reversal is used to compensate for incorrectly wired balanced cables, to prevent signal cancellation when a source is dual-miked from opposite directions, or to reverse left/right with microphones in a mid-side (MS) configuration.

Phantom Power

Phantom powering is a fixed DC voltage of either 12 or 48 Volts. This voltage is resistively applied to pin 2 and pin 3 of an input's XLR connector, relative to pin 1. In this configuration, there is no voltage difference between signal pins 2 and pin 3.

The phantom voltage is selectable between 12 and 48 Volts from the Setup Menu item **INPUTS > Phantom Voltage**. The selected voltage level applies to all inputs with phantom power enabled. The factory default phantom power voltage is 48 V.

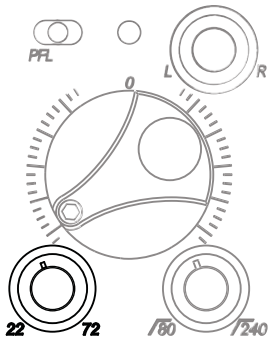
Phantom power can be activated for each input. To enable phantom power, enter the input's channel screen, press the Headphone Controller, highlight either MIC-PH or LINE-PH, press Headphone Controller again to make the selection.

Gain/Trim and Fader Relationship

The gain of an input is adjusted by two controls, Input Trim and Input Fader. This two-stage architecture is identical to the topology of large mixing consoles and provides a great deal of control. Input Trim is often thought of as a course gain control and the Input Fader as the fine gain control.

Input Trim (Analog)

The 664's analog input sensitivity is set with the pop-up Trim control knob. With the Input Fader set to unity gain (0 dB or 12 o'clock), make the appropriate adjustments using the Trim control. Once the coarse gain is set to the desired level, push the Trim control to hide it from the 664's mixing surface. Analog trim level is adjustable from +22 to +72 dB of gain.

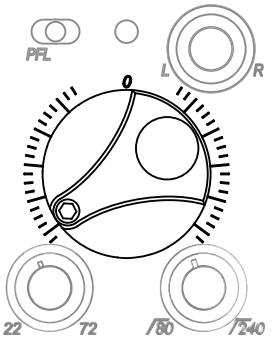


Input Trim (Digital)

The coarse level of a digital input is set by rotating the Select Encoder in the Input Settings Screen of an input with a digital source. The digital trim level is displayed on the LCD when adjusted. The digital trim level is adjustable from -20 to +50 dB.

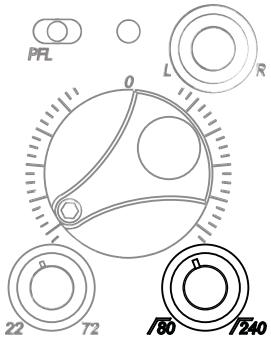
Input Fader

The Input Fader is the primary control used while mixing and it affects the level of the Input signal routed to all post-fade destinations. Use the Input Fader to make fine gain adjustments. The fader can be attenuated from off (full counter-clockwise position) to +15 dB above the set Trim level (full clockwise position). To optimize gain structure for the best performance, operate input faders at or near the 0 dB (unity gain) position.



High-Pass Filter

Each input channel has an adjustable high-pass filter controlled by the High-Pass Filter control. High-pass (or low-cut/low roll-off) filters are useful for removing excess low frequency energy from audio signals. Wind noise is a common unwanted low frequency signal that can be reduced with the use of a high-pass filter. For most audio applications, engaging the high-pass filter is beneficial, because audio information below 100 Hz is rarely used, especially for speech reproduction.



The 664's high-pass filter circuit features an adjustable corner (-3 dB) frequency over a range from 80 to 240 Hz. Below 80 Hz, the filter's slope is 12 dB/octave. At higher corner frequency settings, the slope is 6 dB/octave. The purpose for this compound slope is to give additional roll-off at the 80 Hz setting to reduce wind noise and low frequency rumble. The higher settings can be used to counteract the proximity effect of directional microphones where a more gentle slope is desired.

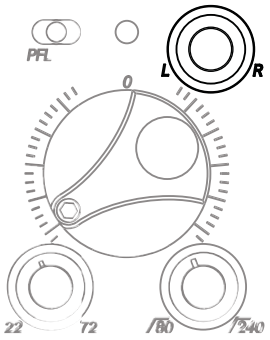
The 664's high-pass filter circuit is unique because of its placement before any electronic amplification. Most mixers' high-pass filter circuits are placed after the microphone preamplifier, such that all of the low-frequency signals get amplified. By virtue of the 664's circuit cutting the low-frequency signals before amplification, higher headroom is achieved in the presence of signals with significant low-frequency energy.

When possible, attempt to equalize at the sound source with microphone selection, placement, windscreens, and onboard microphone filtering. Many microphones have on-board high pass filters. Use the high-pass filters on the 664 in conjunction with the microphone's filter to increase the filter's slope.

The filter can be removed from the circuit completely by rotating the high-pass filter control to the full counter-clockwise (detented) position. The high-pass filter potentiometer can be adjusted easily and then recessed to hide it from the mixing surface.

Pan Control

The pop-up Pan Control routes inputs to the left (L) and right (R) channels of the stereo Master Bus. The pan pot has a detent in the center position. After setting the pan, the pan control can be recessed to hide it from the mixing surface during normal operation.



Input Linking

Input pairs 1-2, 3-4, and 5-6 can be linked as stereo pairs. When a pair of inputs is linked:

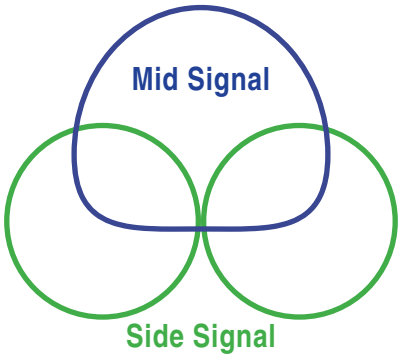
- Each channels' Trim Control and High-Pass Filter Control work as normal, controlling coarse gain and high-pass filtering for their respective inputs.
- The odd channel's Input Fader controls the post-fade level of both inputs.
- The odd channel's Pan Control controls the balance of the stereo signal to the Master Bus. When linked MS, the odd channel's Pan Control functions as a left/right balance control for the matrixed MS signal.

- The even channel's Fader Control and Pan Control are disabled.
- The limiters of both inputs are linked.
- The background label of both inputs is connected on the Main Screen.

M/S Matrixing

When input pairs are linked MS, the odd channel is used for the Mid signal and the even channel is used for the Side signal. To produce a stereo signal from an M/S configuration, the signal from both microphones must be processed.

Mid-side (MS) matrixing is a method for processing audio signal from a cardioid microphone and a bi-directional microphone into a stereo signal. The cardioid microphone is the "mid" signal and connects to the odd Input, and the bidirectional microphone is the "side" signal and connects to even Input. The cardioid microphone is pointed at the sound source, and the bidirectional microphone is oriented sideways (positioned with its capsule as near as possible to the cardioid microphone's capsule). The following diagram shows the relative polar patterns of microphones in an M/S configuration.

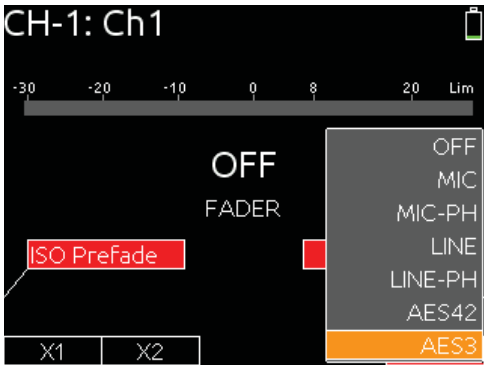


Digital Inputs

The 664 accepts AES3 (AES/EBU) balanced, AES42 (digital microphone) balanced, and AES3id unbalanced digital signals on the Input 1 and Input 6 XLR connectors. The 664 auto-detects between AES3 and AES3id digital signals and adjust accordingly. Digital input gain is controlled from the front panel faders.

Never attach unbalanced connections to an input set for AES42. This can result in damage to the hardware.

To use a digital input, slide the Input 1 or Input 6 Input Selector Switch to the right to enter the Input Settings Screen. Press the Headphone Encoder to display the list of available input sources. Turn the Headphone Encoder to select AES3 or AES42, and press the Headphone Encoder to confirm the selection.



Each connector carries two channels of digital audio. Alternate channels of digital audio sent to Input 1 or Input 6 are available on adjacent inputs, according to the following table.

AES Input Channel:	Available to 664 Input:
XLR Input 1, Left Channel	Input 1
XLR Input 1, Right Channel	Input 2 (Available when Input 1 source is set to AES)
XLR Input 6, Left Channel	Input 5 (Available when Input 6 source is set to AES)
XLR Input 6, Right Channel	Input 6

The 664's digital inputs are sample rate converted to the internal recorder's sample rate. To sync the sample rate of the 664's internal recorder to external digital audio devices, *see Sampling Rate*.

Things to consider when using AES digital inputs:

- When using an unbalanced AES3 (SPDIF) input, the other available digital input should not be used with an AES42 microphone.
- Signal from digital inputs is not available on direct outputs.

Limiters

Limiters prevent clipping by attenuating signals that surpass a set threshold. The amount of attenuation is defined by the "ratio" of the limiter and expressed as two numbers. All 664 limiters use a 20:1 compression ratio. This means that signal that exceeds the threshold by 20 dB will exit the limiting stage at only 1 dB above the threshold.

The time it takes for limiting to begin once signal has exceeded the threshold is referred to as the "attack time" and the time it takes for limiting to cease once signal has fallen back below the threshold is referred to as "release time". Signals that exceed the threshold faster than the limiter's attack time can still cause clipping. The 664 limiters have a 1 ms attack time and a 500 ms release time.

The limiters are globally activated when the Setup Menu item **LIMITERS > Limiters** is set to **On**. This activates both the input limiters (Inputs 1 through 6) and all output limiters. Sound Devices recommends using the limiters at all times. Limiters are present on both mic and line-level inputs as well as the Master L,R tracks and the X1,X2 tracks. The 664 input limiters have a threshold of +16 dBu (4 dB below clipping), while the limiters for the L, R, X1, and X2 tracks are adjustable from the Setup Menu item **OUTPUTS > L,R Limiter Thresh** and **OUTPUTS > X1,X2 Limiter Thresh**.

In normal operation, with a properly set gain structure, the threshold of the Input Limiter is rarely reached. Without Input Limiters, high signal conditions can overload a channel and cause distortion. The Input limiter is working when the respective input's Input Activity LED illuminates yellow. If the

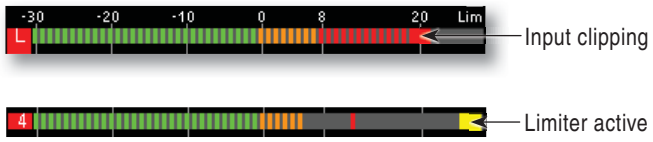
Activity LED is regularly in the yellow, reduce the amount of gain applied to the channel by turning down the Trim control. *See Input Activity LED for additional information.*

When Inputs are linked as a stereo pair, the Input Limiters are also linked and perform the same gain reduction equally to both inputs.

The Output Limiters prevent the L,R and X1,X2 signal from exceeding the user-set limiter threshold. Setup Menu options **LIMITERS > L,R Limiter Thresh** and **LIMITERS > X1,X2 Limiter Thresh** allow limiter thresholds to be set set in 1 dB increments from +4 dBu to +20 dBu.

Metering

All track levels are displayed on the LCD in the Main Screen. RTN input levels are displayed on the RTN Screen (Press the Meters button to toggle between the RTN Screen and the Main Screen). Each segment of the meter represents 1 dB. A larger red square under the 20 dBU marker indicates clipping on the track. A large yellow square indicates limiting activity on the track.



Meter Ballistics

VU

VU (volume units) meter ballistics correspond closely to how the human ear perceives loudness. This provides a good visual indication of how loud a signal will be. In VU mode, the attack and decay of the meter signal is 300 mS. VU meters provide good visual indication of how loud a signal will be, but provide poor information of actual signal peaks.

Peak + VU

The 664 can simultaneously display VU and Peak level information. In this mode the perceived loudness (VU) is displayed as a standard bar, and the Peak signal as a single, independent segment above the VU.

Peak Only

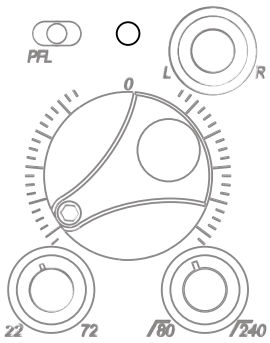
Peak-reading ballistics (PPM) correspond to actual signal peaks, but don't necessarily correspond to perceived signal loudness. Peak meters have an instantaneous attack and a slow decay to allow visual monitoring of peak activity. Peak metering is useful when interconnecting to audio inputs on digital equipment. In the digital realm, signal overload can cause immediate distortion.

Peak Hold

When Peak Metering is enabled (Peak Only or Peak + VU), peak hold displays the last highest peak value on the meter as a separate, individual meter segment. By default this meter segment will remain visible for 1 second. This time can be adjusted to 1, 2, 3, 4, or 5 seconds. The Infinity option will display the last highest peak indefinitely until a higher peak is reached. The Off option will disable peak hold. The Peak Hold setting applies to meters on the Main Screen and in all Input Settings windows.

Input Activity LED

Each Input has its own Input Activity LED located just above the Input Fader. The LED illuminates in various colors and intensities to represent the signal level appearing at its respective input. Green = pre-fade signal activity, yellow = pre- and post-fader limiter activity, red = pre- and post-fader signal overload (peaking). Reduce the trim level control if the LEDs continuously illuminate yellow or red. The Input Activity LED will flash yellow when the Input PFL is latched. *see PFL.*



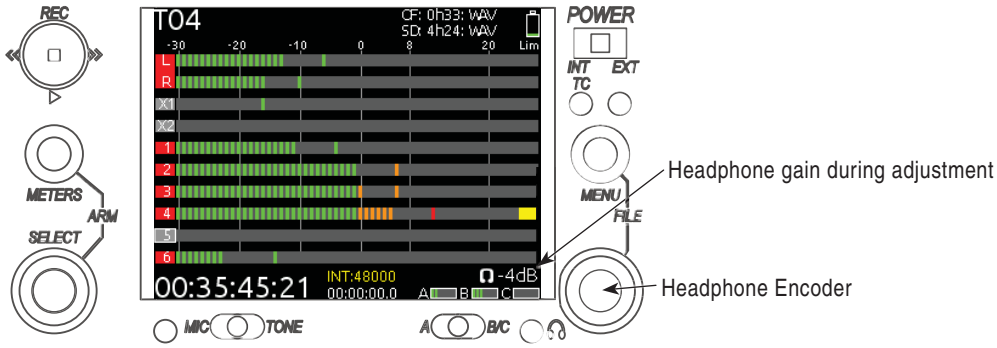
Headphone Peak LED

The Headphone Peak LED is located just left of the Headphone Encoder. This LED will illuminate red to indicate clipping in the headphone amplifier. Monitoring without a visual indication of headphone clipping can mislead the sound mixer into thinking that the output or return feeds are distorted.

Headphone Monitoring

Headphone Gain

Headphone output level is controlled by turning the Headphone Encoder while viewing the Main Screen or Input Settings Screen. When adjusting the headphone gain, the gain value (in dB) is displayed in the lower right corner of the LCD (Near the Headphone Encoder). This space displays the currently selected headphone source when the headphone gain is not being adjusted.



The 664 can drive headphones to dangerously high volumes. Turn down the headphone gain before selecting a headphone source to prevent accidental signal extremes.

Headphone Source Selection

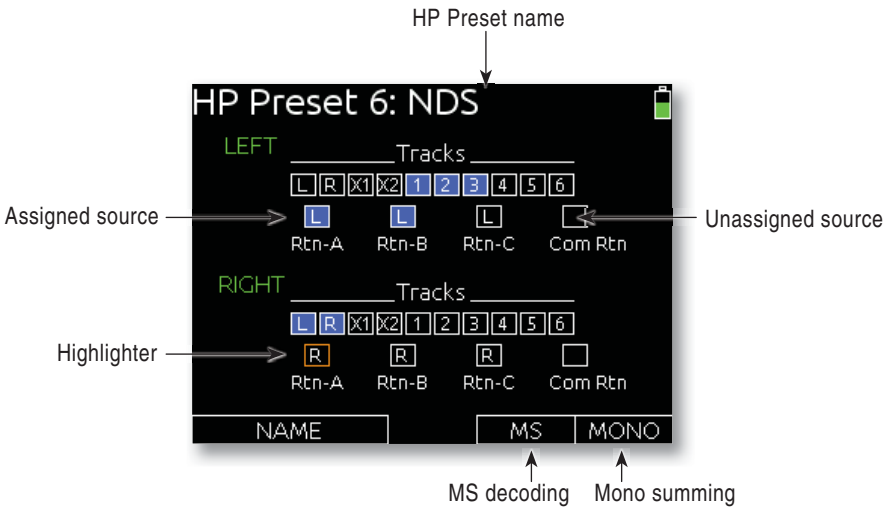
To select the monitor source, push the Headphone Encoder while viewing the Main Screen to display the Monitor Source List. This will display a list of available monitor sources. Turn the Headphone

Encoder to highlight a source, and press the Headphone Encoder to select that source. The signal to the headphone outputs will immediately change to the selected program.

Monitor Source	Description
LR ST	Master Bus in stereo.
LR Mono	Master Bus summed mono to both ears.
L Mono	Left channel of Master Bus sent to both ears.
R Mono	Right channel of Master Bus sent to both ears.
LR MS ST	Mid-Side Stereo - Master Bus decoded MS stereo to headphones, this is not to be used if the inputs are already linked as an MS pair.
X1X2	Aux Bus in stereo.
HP Preset	HP Presets 1-10 are customizable monitor sources. *See Headphone Presets*

Headphone Presets

Ten custom monitor sources are available. These headphone presets can be edited, named and saved. To edit a custom headphone preset, press the Headphone Encoder to access the Monitor Source List, highlight the desired preset, then slide the Slate/Tone Switch. This displays the Headphone Preset Screen:



Available sources are displayed for each monitor channel (left and right). Sources with a blue background are assigned to that channel of the headphones, and sources with a black background are not assigned to the headphone channel. To toggle the assignment of a source, turn the Headphone Encoder to highlight the source and press the Headphone Encoder to toggle the assignment.

Slide the RTN Switch left to toggle MS decoding for the preset. MS decoding will decode the entire preset and should not be used if inputs are already linked MS. Slide the RTN Switch right to toggle-mono summing of the preset. Mono summing will ignore left and right assignment and put all active sources in both channels of the preset.

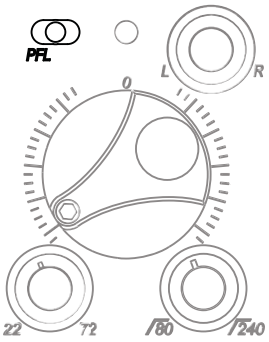
When the Headphone Preset is edited, it can be named so it can be easily identified in the Monitor Source List. To name a Headphone Preset, slide the Slate / Tone Switch either left or right while viewing the Headphone Preset Screen.

Headphone Tones

The start of a recording is indicated audibly by a single 440 Hz tone sent to headphones. When recording is stopped two 220 Hz tones are sent to the headphones. The Record/Stop Bell can be enabled from the Setup Menu item **SYSTEM > Record/Stop Bell**. The Setup Menu option **SYSTEM > Warning Bell Level** allows adjustment of the level of the Record/Stop Bell.

PFL (Channel Solo Monitor)

Slide the PFL switch left to activate, and again to deactivate. For momentary action, hold the switch left for one second or longer.



PFL stands for "Pre-Fade Listen". When any input's PFL is activated, the currently selected Monitor Source is replaced with the PFL mix program. The PFL mix program is a mono mix of the pre-fade level of all inputs with PFL enabled. PFL monitoring only affects the headphone monitor, it does not affect audio sent to the outputs.

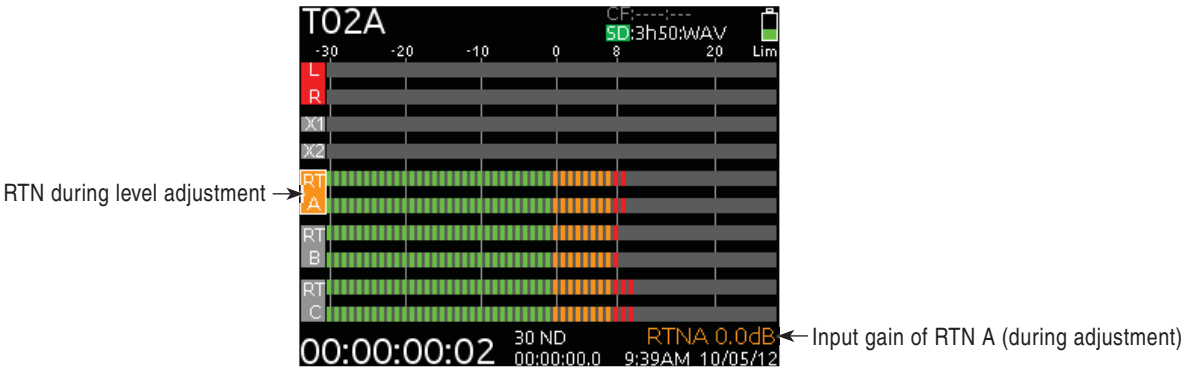
The selected input's Activity LED flashes yellow when the input's PFL is engaged. When one input PFL is engaged, that input's name will appear in the lower right corner of the Main Screen. When multiple input PFL's are engaged, **PFL MULT** will appear in the lower right corner of the Main Screen.

RTN and COMM Monitoring

RTN A, RTN B, RTN C, and COM RTN inputs can be monitored quickly using the RTN Switch. Primary Monitor Sources are monitored by sliding the RTN Switch left or right while Alternate Monitor Sources are monitored by holding the Select Encoder then sliding the RTN Switch left or right. The Primary and Alternate Monitor Sources can be changed from the Setup Menu options **COMMS/RETURNS > RTN Toggle Left** and **COMMS/RETURNS > RTN Toggle Right**. To return to the previous monitor selection, Slide the RTN Switch in the same direction of the active RTN.

RTN A, RTN B, and RTN C input levels can be metered and adjusted from the RTN Screen. To access the RTN Screen, press the Meters Button when viewing the Main Screen. To adjust the gain of a RTN input:

1. Access the RTN Screen, if it is not visible already.
2. Turn the Select Encoder to highlight the RTN to be adjusted.
3. Press the Select Encoder to access gain adjustment mode for the selected RTN. The background color of the RTN label will become orange, and the input gain value (in dB) will appear in the lower left corner of the LCD.
4. Turn the Select Encoder to adjust the gain. RTN gain adjustment mode will exit automatically after 3 seconds.



RTN monitoring will cancel the PFL mix program when activated.

Output Setup and Control

The 664 features a host of analog and digital outputs to accommodate for complex, multi-camera set-ups. The 664 has 2 stereo output busses: The Master Bus (LR), and the Aux Bus (X1/X2).

Master and Aux Outputs

There are 4 pairs of balanced analog output connections for the Master Bus (XLR, TA3, and 2 Hirose 10-pin), and one pair of balanced analog output for the Aux Bus (TA3). Additionally, there are two stereo unbalanced outputs for the Master Bus (TA3 and 3.5mm Tape Outputs).

The XLR and both Hirose 10-pin outputs are transformer-balanced, each driven from their own transformer windings for excellent isolation. Each output can be independently set to Line (+4 dBu nominal), -10, or Mic level (40 dB of attenuation versus Line) from the Setup Menu section **OUTPUTS**. The master outputs are capable of driving long cable runs.

XLR

By default the XLR outputs send a balanced analog signal. These connections can be independently configured to send AES3 signal. *see AES Digital Outputs for details.* When set to analog output, the L and R XLR connections always carry the Left and Right Master Bus program.

TA3 (Master L, R, X1, and X2)

The TA3 outputs send a balanced analog signal. The X1 and X2 TA3 connections always carry X1 and X2 Aux bus program. The L and R TA3 connections always carry the Left and Right Master Bus program.

Hirose 10-pin

Each Hirose 10-pin connection has a balanced pair of outputs that carry the Left and Right Master bus program. By default, each of the balanced outputs on the Hirose 10-pin connections sends a balanced analog signal. Each of the balanced outputs on the 10-pin A connections can be independently configured to send AES3 signal. *see AES Digital Outputs for details.*

Each Hirose 10-pin also includes an unbalanced stereo return input for headphone monitoring. The Sound Devices XL-10 Breakout Cable is an available accessory that provides easy access to the balanced outputs and stereo returns of the 10-pin Hirose connections.

Direct Outputs

Inputs 1-6 have corresponding direct outputs on balanced TA3 connections. By default, these outputs are pre-fade and line level. The direct outputs can be set to post-fade from the Setup Menu item **OUTPUTS > Direct Out Pre/Post**. The direct outputs can be set to Line (+4 dBu nominal), -10, or Mic level (40 dB of attenuation versus Line) from the Setup Menu item **OUTPUTS > Direct Out Levels**.

AES digital input signals are not available on direct outputs.

Tape Output

The 664 includes two unbalanced tape level outputs, a TA3 connector and a 3.5 mm jack. These two connectors are resistively in parallel and are isolated from the balanced outputs.

Tape Outputs, by default, send the Left and Right stereo program. The Tape outputs can be used to send the incoming RTN A, RTN B, or RTN C signals directly to receiving devices from Setup Menu

option **OUTPUTS > Tape Out Source**. This is useful for sending an IFB wireless feed of the RTN audio to the producer for video playback.

Master and Aux Bus Level

The output gain of the Master Bus and the Aux Bus is adjustable from Off, -30 to 0dB. To adjust output gain:

1. From the Main Screen, turn the Select Encoder to highlight the output(s) to be attenuated.
2. Push the Select Encoder. The output level of the highlighted track will be displayed in the lower right-hand corner of the LCD. *If no adjustment is made before 3 seconds, the selection will cancel. This is to avoid accidental adjustments.*
3. Turn the Select Encoder to adjust the gain.

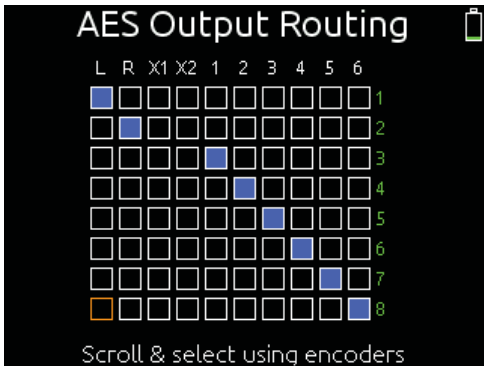
Outputs that are linked (from Setup Menu item **OUTPUTS > Linking**) will be shown as a connected box on the Main Screen and their gain will be adjusted simultaneously.

AES Digital Outputs

There is a total of 8 available channels of digital output on 4 connections. Each of the XLR or 10-pin A outputs can be configured to output AES3 digital signals from the Setup Menu options **OUTPUTS > XLR-R Out**, **OUTPUTS > XLR-L Out**, and **OUTPUTS > 10-pin A Out**.

Any track can be routed to any AES output in any combination. The AES output routing can be edited from the Setup Menu item OUTPUTS > AES Output Routing. This interface consists of a matrix in which the columns are tracks and the rows are the numbered AES outputs. The blue boxes indicate routed tracks and the black boxes indicate unrouted tracks. Turn the Select Encoder to move the orange highlighter vertically and turn the Headphone Encoder to move the highlighter horizontally. Push either the Select Encoder or the Headphone encoder to toggle the routing status of the highlighted track.

The following screen shot shows the AES Output Routing screen. The highlighter is in the lower-left corner.



COM Setup

The 664 features an unbalanced stereo COM Send (TA3) and an balanced mono COM RTN (TA3) for communications purposes. These special input and output channels can be used to privately communicate with a boom operator or other member of the production crew.

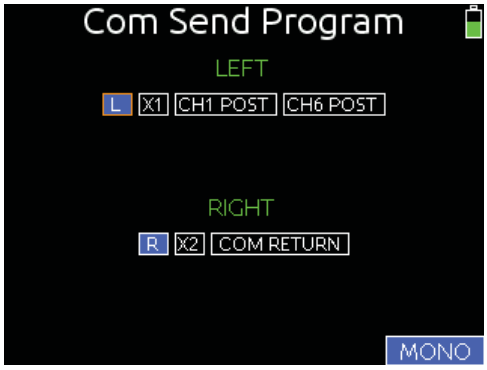
During operation, it is possible to interrupt the Com signal and speak to the boom operator via the slate mic by Holding down the Select Encoder and sliding the Slate/Tone Switch to the left. The green Slate LED will illuminate to indicate that the slate mic is routed to the Com Send. During this time, the slate mic signal is sent only to the Com Send Output and right Headphone output, but not to record tracks or any other outputs. To cancel slate communication and resume the Com Program, slide the Slate/Tone Switch to the left again. The green Slate LED will turn off.

The gain of the Com Send and Com Return can be adjusted from the Setup Menu options **COMMS/RETURNS > Com Send Gain** and **COMMS/RETURNS > Com Return Gain**, respectively.

COM Send Program

By default, the program sent to the COM Send output is the Left and Right Stereo Bus. To change the program of the COM Send:

1. Enter the Setup Menu and select **COMMS/RETURNS**.
2. Select the **Com Send Program** Option.
3. Turn the Headphone Encoder to select the desired source, and push the Headphone Encoder to toggle the assignment of the source to the Left or Right side of the COM Send.
4. Optionally, slide the RTN switch to toggle mono or stereo of the COM Send.



Com Program Auto-Mute

By default, the Com Send program will mute when slate communication is engaged. This helps the boom operator hear the voice of the 664 operator. This behavior can be disabled so that the Com Send program is not muted when communication is engaged via the Setup Menu option **COMMS/RETURNS > Program Auto-Mute**.

Slate/Com Mic & Tone Oscillator

The 3-position Slate / Tone Switch controls both the slate microphone and the tone oscillator.

Slate Microphone

Push the Slate / Tone Switch left to activate the slate mic. The green Slate Mic LED will illuminate to indicate that the slate mic is active. Push the Slate / Tone Switch left again to deactivate the slate mic. Hold the Slate / Tone Switch left for 1 second to activate the slate mic momentarily (until the switch is released).

The secondary slate function is accessed by holding down the Select encoder when using the Slate / Tone Switch. By default, the secondary slate function activates the slate mic to the COM Send (*see COM Setup*). The primary and secondary slate functions can be swapped from the Setup Menu option **COMMS/RETURNS > Mic Toggle Switch**.

Program audio is muted when the slate mic is activated. This behavior can be disabled (so that program audio does not mute when the slate mic is activated) from the Setup Menu option **COMMS/RETURNS > Program Auto-Mute**.

Internal Slate Mic

The default slate mic is the 664's internal slate microphone. The 664's built-in slate microphone is used to notate scenes from the mixer location. Its audio performance is not suitable for critical recording applications; it should be used for documenting scenes and for communication purposes only.

External Slate Mic

An external microphone can be used instead of the built-in slate microphone. To use an external slate mic, plug the mic into the TA3 Slate Mic input on the Left panel and adjust Setup Menu option **COMMS/RETURNS > Slate/Com Mic Source** to **Ext Mic** (for dynamic microphones) or **Ext 12V Mic** (for condenser microphones).

Tone Oscillator

Push the Slate / Tone Switch right to activate the tone oscillator. Hold the Slate / Mic Switch right for more than 1 second to lock the tone oscillator on. When the tone oscillator is locked on, push the Slate / Mic Switch right again to deactivate.

By default, tone will be sent to the Master Bus (L,R), its associated outputs, and all direct outs. Tone can additionally sent to the X1 and/or X2 tracks. Access Setup Menu option **SYSTEM > Tone Routing** to adjust this.

Alternate Slate/Com Switch

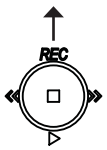
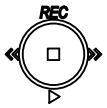
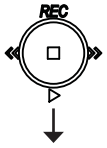
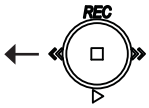
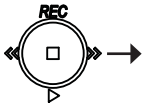
By default, COM communication is engaged by sliding the Slate/Tone Switch to the left while holding the Select Encoder and normal slate operation is engaged by simply sliding the Slate/Tone Switch to the left. This functionality can be reversed from the Setup Menu option **COMMS/RETURNS > Mic Toggle Switch**.

Digital Audio Recorder

The 664 features a powerful multi-track digital recorder. The internal recorder can record up to 10 tracks (16 tracks with the optional CL-6 Input Expander. *see CL-6*) of uncompressed PCM audio (.WAV file extension) or 2 tracks of compressed MP3 audio (.MP3 file extension). *See File Formats.* Data is written to removable CompactFlash (CF) and SD/SDHC/SDXC (SD) cards. *See Record Media.*

Transport Control

The 5-position Transport Control is used to perform all recording and playback functions.

Function	Direction	Action
Record		**Push the Transport Control Up** Begins recording a new file.
Pause / Stop		**Press the Transport Control** During Record, press once to stop recording. During Playback, press once to pause, press again to stop. In standby, hold to display next take name.
Play		**Push the Transport Control Down** Begins playback of the last file recorded or file currently loaded. *See Playback*
Rewind/ Load Previous Take		**Push the Transport Control Left** While in Stand-by, push left to load the previous take. While in playback, push and hold left to rewind.
Fast Forward/ Load Next Take		**Push the Transport Control Right** While in Stand-by, push right to load the next take. While in playback, push and hold right to fast forward.

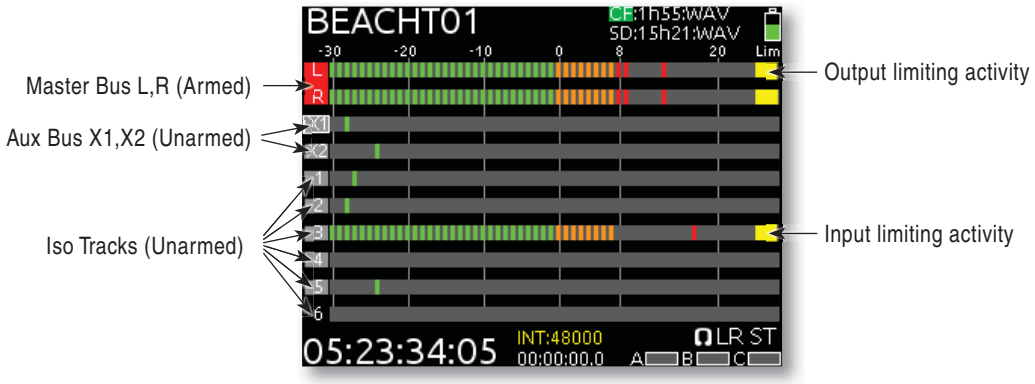
Recording Tracks

Each input of the 664 has a corresponding Iso Track and is permanently routed to it. Inputs 1-6 can be routed to their respective ISO tracks pre- or post-fade (*see Input Setup and Control*). Four additional record tracks are available which correspond to the stereo Master Bus (L,R) and the stereo Aux Busses (X1,X2) *See Bus Assignment for additional information*.

CL-6 inputs 7-12 are routed to their respective ISO tracks post-fade only.

Track Arming

All tracks along with their signal levels, are visible on the Main Screen.



Tracks ready for recording (armed tracks) have a red label. Tracks not assigned for recording (disarmed tracks) have a gray label. To toggle the armed status of a track, turn the Select Encoder to highlight the track, press and hold the Meters Button, then push the Select Encoder. The background color of the track will change to indicate the new arming status. If an input is off (Trim Control fully counter-clockwise or Input Selection set to Off), the track label will be black.

Master Bus and Aux Bus meters are not shown on the Main Screen when the CL-6 Input Expander is attached. See CL-6 Input Expander

Track-to-Media Routing

The 664 supports simultaneous recording to CF and SD media. *See Recording Media for details.* By default, all armed tracks are recorded to both cards as a polyphonic WAV file. Recording options are set independently for each memory card from the Setup Menu items **RECORDER > Record to CF** and **RECORDER > Record to SD**. Each item can be set to the following settings:

Setting	Tracks Recorded	File(s) Generated
Off	None (0 tracks)	No file
Wav Poly	All armed tracks (0-16 tracks)	Polyphonic WAV file
Wav Poly (ISOs only)	Armed ISO tracks (0-12 tracks)	Polyphonic WAV file
Wav Poly (LR only)	Armed L and R tracks (0-2 tracks)	Polyphonic WAV file
Wav Poly (X1X2 only)	Armed X1 and X2 tracks (0-2 tracks)	Polyphonic WAV file
MP3 (LR)	Armed L and R tracks (0-2 tracks)	2-track MP3 file
MP3 (X1X2)	Armed X1 and X2 tracks (0-2 tracks)	2-track MP3 file

Track counts shown are the maximum tracks possible with the CL-6 Input Expander attached. See CL-6 Input Expander

Track Naming

Track names are stored in the metadata of each recorded file. The following list displays the default track names:

- Track L: MixL
- Track R: MixR
- Track X1: Aux1
- Track X2: Aux2
- Track 1: Ch1
- Track 2: Ch2
- Track 3: Ch3
- Track 4: Ch4
- Track 5: Ch5
- Track 6: Ch6
- Track 7: Ch7
- Track 8: Ch8
- Track 9: Ch9
- Track 10: Ch10
- Track 11: Ch11
- Track 12: Ch12

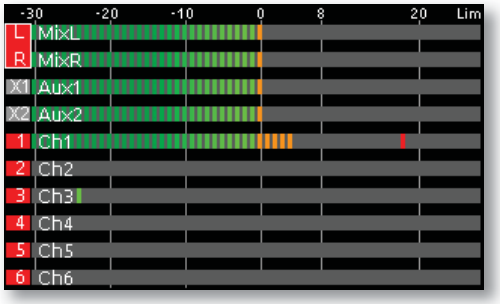
Track names of previously recorded files and track names of all subsequently recorded files can be edited. Tracks can be renamed from the Setup Menu or the Input Settings window. To rename tracks from the Input Settings window:

1. Enter the Input Settings window of the Track to be renamed by sliding the Track's Input Selector Switch to the right and then releasing it.
2. From the Input Settings window, Slide and hold the Input Selector Switch to the right and hold it for 1 second.
3. Using the onscreen keyboard or attached USB keyboard, enter the Track's name.

To rename tracks from the Setup Menu:

1. Enter the Take List (Press and hold the Menu Button then press the Headphone Encoder or **FILE STORAGE > Take List**)
2. To edit track names of subsequently recorded files, highlight the next take (indicated by orange text and the phrase "[next]") and slide the RTN Switch. To edit the track names of a previously recorded take, highlight that take.
3. Slide the RTN Switch to edit the metadata. Rotate the Headphone Encoder to select the track to be renamed, and press the Headphone Encoder to edit the track name.

Track names can be displayed on the Meters of the Main Screen and RTN Screen. To enable track names on the meters, change Setup Menu option **System > Track Names in Meters** to **On**. The following image shows the Main Screen with Track names.



Sampling Rate

The 664's digital audio recorder can record at the following sampling rates:

- 44.1 kHz
- 47.952 kHz
- 47.952 kHz "F" (files stamped 48 kHz)
- 48 kHz
- 48.048 kHz
- 48.048 kHz "F" (files stamped 48 kHz)

The sampling rate of the digital recorder is set from Setup Menu item **RECORDER > Sample Freq**. *See Time Code section for details of "F" modes.* The current Sample Rate is displayed on the Main Screen.

Word Clock Synchronization

By default the 664 recorder uses its own internal clock to generate its sampling rate. The BNC Word Clock input and Word Clock output on the rear of the 664 can be used to clock the 664 from an external word clock signal or to synchronize external audio devices to the 664's word clock.

To synchronize the 664 to an external word clock signal, connect the word clock signal to the Word Clock Input BNC on the rear of the 664 and change **TIMECODE/SYNC > Sync Reference** to **Word Clock**.

When the 664 is locked to external word clock, it is indicated on the Main Screen:

When Setup Menu item **TIMECODE/SYNC > Sync Reference** is set to **Word Clock**, but no valid external word clock is present, it will be indicated on the Main Screen:

The 664 only accepts word clock signals between 44.1 kHz and 48.048 kHz.

Bit Depth

When recording WAV files, the 664 records either 16 or 24 bit files. Bit depth is selected from the Setup Menu item **RECORDER > Bit Depth**.

Bit Depth and Dynamic Range

Bit depth defines the digital "word length" used to represent a given sample and correlates to the maximum dynamic range that is represented by the digital signal. Larger bit depths accommodate more dynamic range. Sound Devices recommends 24-bit recording for all critical production.

Recording Media

The 664 records to approved removable CompactFlash (CF) and SD/SDHC/SDXC (SD) cards. The 664 can write to one or both of these cards at the same time. Visit *www.sounddevices.com/approved* for an up-to-date list of tested and approved cards.

The SD card is inserted with the bottom of the card (the side with metal contacts visible) facing toward the rear of the 664. Push until it clicks. To remove, push in again to release the spring-loaded connector, then pull the SD card out.

The CF card is inserted with the top of the card (the side with the manufacturer's branding label) facing the rear of the 664. The final few millimeters of insertion require light force. To remove the CF card, grip it firmly and pull it out.

If the CF card is oriented correctly, it will slide in easily until only a few millimeters are protruding. If the CF card is oriented incorrectly, it will slide in easily until over 1 cm of the card is still protruding.

Formatting Media

The 664 uses the FAT32 file system. FAT32 is compatible with all major operating systems. Before recording to CF or SD media, cards must be formatted. To format cards in the 664:

1. Press the Menu button to enter the Setup Menu.
2. Turn the Headphone Encoder to highlight **FILE STORAGE**, then press the Headphone Encoder to enter the **FILE STORAGE** section.
3. Turn the Headphone Encoder to highlight **Erase/Format CF** or **Erase/Format SD**, then Press the Headphone Encoder to begin the formatting process.

Time Code

The 664 holds accurate time code for up two hours between battery changes using its internal, Li-Ion time code battery. After two hours without power, the 664 reverts to a slightly less-precise time-of-day crystal to maintain the date/time of the unit. This time code battery is charged whenever the 664 is powered up.

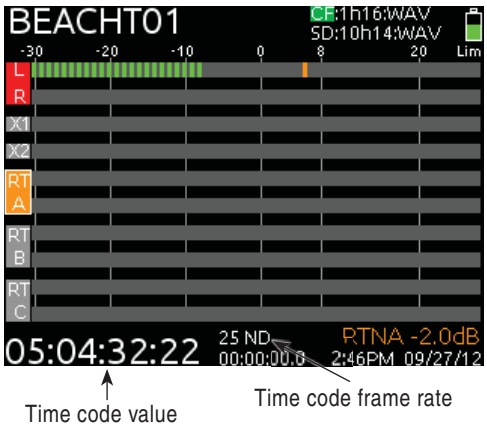
If the time-of-day clock is reset during the production day or if the time code mode is changed from 24 hour run to another mode and back, the time code value will change. You must re-jam all time code devices to ensure proper synchronization.

File-based recorders place a time code and frame rate stamp in the BEXT and iXML chunks of an AES31 (Broadcast WAV) file. During playback, the 664 generates SMPTE time code from this number and extrapolates it based on the time code frame rate.

All files generated by the 664 have time code numbers that begin on the 0 frame (or 02 in DF modes) and end on the 0 frame such that a file's duration is always an integer number of seconds long. If necessary, pre-roll and post-roll is dynamically applied to accomplish this. This simplifies synchronization in post-production.

Sound Devices Wave Agent Beta for Mac OS and Windows computers allows users to add or edit Beginning Time Code Stamps. *See Wave Agent for more details.*

The time code value and frame rate of the 664 are displayed on the RTN Screen:



Timecode modes

The time code mode determines if the 664 generates or reads timecode and when time code runs and stops. Time code mode is changed from the Setup Menu option **TIMECODE/SYNC > Timecode Mode**.

Rec Run (Record Run)

Generator mode. The 664 sends running timecode while recording and stationary timecode while not recording. Timecode in this mode defaults to the last stationary value at power-up. When switching to Record Run from another mode, the internal generator will stop at the last number generated. A user-defined value can be jammed into the internal generator from the Setup Menu option **TIMECODE/SYNC > Set Generator TC**.

Free Run

Generator mode. Timecode runs continuously. The value can be set by "jamming value" in the Setup Menu option **TIMECODE/SYNC > Jam Menu** or by manually setting a value from the Setup Menu option **TIMECODE/SYNC > Set Generator TC**.

24h Run (24 hour Run)

Generator mode. Timecode runs continuously. When this Time Code Mode is selected or when the time/date is altered, the time code value is jammed to the value of the time-of-day clock. The value can be set by "jamming value" in the Setup Menu option **TIMECODE/SYNC > Jam Menu** or by manually setting a value from the Setup Menu option **TIMECODE/SYNC > Set Generator TC**.

Ext-TC (External Time Code)

Reader mode. The internal timecode generator follows an external timecode signal appearing at the time code input. If the external timecode is removed the internal generator stops as well. Recording is triggered manually with the Transport Control.

Ext-TC Auto-Rec (External Time Code Auto-Record)

Reader mode. The internal timecode generator follows an external timecode signal appearing at the time code input. If the external timecode is removed the internal generator stops as well. The 664 automatically begins recording when running time code is sensed. When stopped time code is sensed, recording stops.

Ext-TC/Cont (External Time Code Continuous)

Reader mode. The internal timecode generator follows an external timecode signal appearing at the time code input. If the external timecode is removed the internal generator continues to run to preserve continuous timecode. Recording is triggered manually with the Transport Control.

Ext-TC Auto-Rec/Cont (External Time Code Auto-Record Continuous)

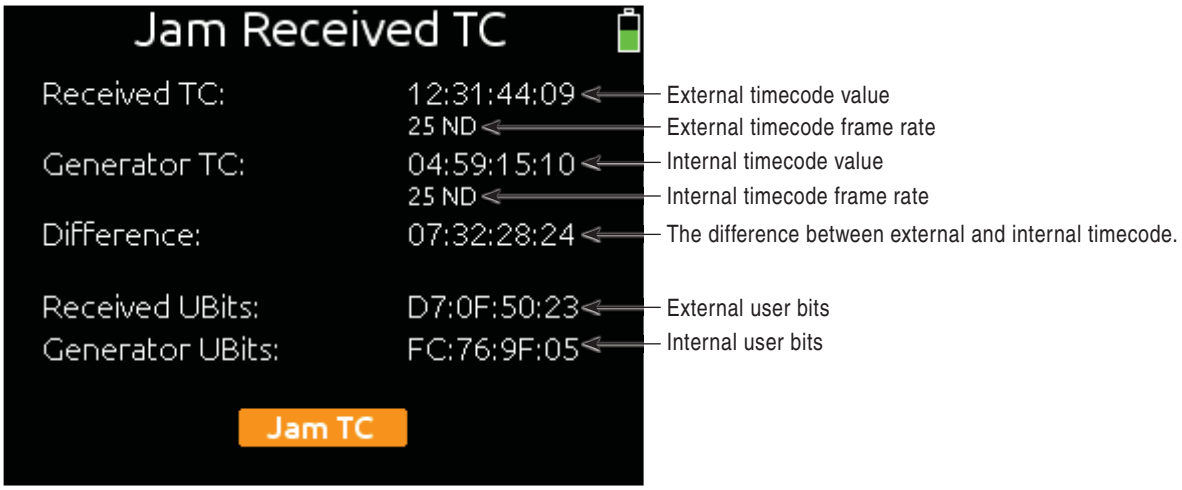
Reader mode. The internal timecode generator follows an external timecode signal appearing at the time code input. If the external timecode is removed the internal generator continues to run to preserve continuous timecode. The 664 automatically begins recording when running time code is sensed. When stopped time code is sensed, recording stops.

Timecode Frame Rate

The frame rate of the time code generator is set with the Setup Menu option **TIMECODE/SYNC > Frame Rate**. When using an external time code mode, ensure that the 664's frame rate is equal to (or cross-jam compatible) the external frame rate.

Jam Menu

The Setup Menu option **TIMECODE/SYNC > Jam Menu** displays the timecode Jam Menu. This special screen displays detailed information about the 664's internal SMPTE timecode generator and SMPTE timecode present on the 664's time code input.



To jam the 664's internal timecode (and User Bits) to external time code, push the Headphone Controller while the Jam Menu is displayed. To leave the Jam Menu without jamming to external timecode, press the Menu Button or the Meters Button.

To manually set the User Bits, use Setup Menu option **TIMECODE/SYNC > Set Generator UBits**.

Timecode Hold Off

Some devices that send rec-run timecode will sometimes have a tendency to send short burst of running timecode. When using timecode to trigger recording on the 664 (External timecode auto-record modes) this behavior can result in unintentional recordings on the 664. To eliminate this problem, the 664 features a Timecode Hold Off function.

When using external timecode auto-record modes, the 664 will wait ("hold off") for the amount of time specified in Setup Menu option **TIMECODE > Hold Off** before beginning recording. Running timecode lasting less than the Timecode Hold Off value will not trigger a recording on the 664.

Things to consider when using Timecode Hold Off:

- This feature only applies in **Ext-TC Auto-Rec** and **Ext-TC Auto-Rec/Cont**.
- If Pre-Roll is set, it will be applied (*see Pre-Roll*). Pre-Roll will only capture audio from the initial detection of a time code signal. If no Pre-Roll is selected, the file will begin after the Hold Off time expires. It is best practice to set Pre-Roll, in the Setup Menu option **RECORDER > Pre-Roll Time**, to a value greater than the selected Timecode Hold Off value. This ensures that audio is captured from the moment a timecode signal is detected and that unintentional files triggered by short bursts of time code are not generated.

F Sampling Rate Modes

48.048k and 48.048kF

The 48.048kF mode (F stands for fake, faux, Fostex—take your pick) is a specific compatibility mode for use with the Fostex DV40 software (1.74 and previous), Avid, Final Cut Pro, and other post-production environments that do not recognize audio files written at 48.048 kHz. In this mode files are

recorded at a 48.048 kHz sampling rate but are stamped at 48 kHz. When played, they will play back 0.1% slower than real time.

One use for the 48.048kF mode is to force a 0.1% speed reduction (pull down) of audio to match MOS-telecined film (24 fps-to-NTSC) in non-linear edit systems, such as Avid or Final Cut Pro. Since the file is stamped as a 48 kHz file, the edit system will play it back at 48 kHz and not at 48.048 kHz. This "audio pull down" will match the transferred picture without the need for an intermediate step through other software to create the pull down.

The time code frame rate (actual recording rate) is forced to 30 ND in 48.048kF mode. The LCD display will show 30 ND during recording. No other frame rate is available in 48.048kF mode.

The audio file, however, is stamped with a 29.97 ND frame rate, along the 48 kHz sampling rate. It will appear as if the file was originally recorded at a 48 kHz sampling rate at a TC rate of 29.97 ND.

Fostex DV40

When using files recorded in the 48.048kF mode with a DV40 with early software, set the DV40 time code frame rate to 29.97ND. Time code stamps will properly match the original time code start times.

47.952k and 47.952kF

Both 47.952 and 47.952kF settings use a record sampling rate of 47.952 kHz, 0.1% lower than 48 kHz. The 47.952kF mode, however, identifies the file as being recorded at 48 kHz. Additionally there are two frame rate options in 47.952kF mode. The time code frame rate can be set to 23.97 while the file is identified as 24ND or it can be set to 29.97ND while the file is identified as 30ND. By default, when 47.952kF mode is selected the frame rate is changed to 29.97ND and identified as 30ND unless the Frame Rate was already set to 23.97 in which case it will remain at 23.97.

Pre-Roll

To prevent missing record cues or up-cutting takes, the 664 has pre-roll buffering available. When active, pre-roll begins recording at a set number of seconds **preceding** the record button being pressed. The Pre-Roll time can be adjust (up to 6 seconds) from the Setup Menu option **RECORDER > Pre-Roll Time**.

Pre-Roll is disabled when the time code mode is set to **Record Run**, **External Time Code Auto Record**, or **External Time Code Continuous Auto Record**. This prevents possible overlapping time code numbers between adjacent files.

Synchronization

A stable sample clock source is essential for high quality digital audio. Setups involving multiple digital recording/playback devices often require that all devices are locked to a common clock reference to ensure they are synchronous, i.e. run at the same speed and in phase with one another. The 664 uses rock-stable word clock and PLL circuitry to generate its internal clock frequency.

The 664 can act as either a clock master (Setup Menu option **TIMECODE/SYNC > Sync Reference** set to **Internal**), where it is the provider of word clock, or as clock slave (Setup Menu option **TIMECODE/SYNC > Sync Reference** set to **Word Clock**), where it locks its internal word clock to a received external master reference clock. External word clock signal is provided via the Word Clock input BNC on the Rear Panel.

When accepting digital input signal, the 664 always converts the sample rate to the recorder's sample rate. During playback, the 664 only uses its own word clock, ignoring external word clock.

File Management

The 664 writes files to both CompactFlash (CF) and Secure Digital (SD) storage media. Storage media must be formatted as FAT32 volumes prior to use with the 664. CF and SD storage media inserted into the 664 can be formatted with Setup Menu option **FILE STORAGE > Erase/Format CF** and **FILE STORAGE > Erase/Format SD** respectively. The FAT32 file system is compatible with all major operating systems. To manage files on 664 media using a computer, simply remove the media from the 664 and insert into a compliant CF or SD card reader connected to a computer.

Visit www.sounddevices.com/approved for an up-to-date list of tested and approved cards.

File List

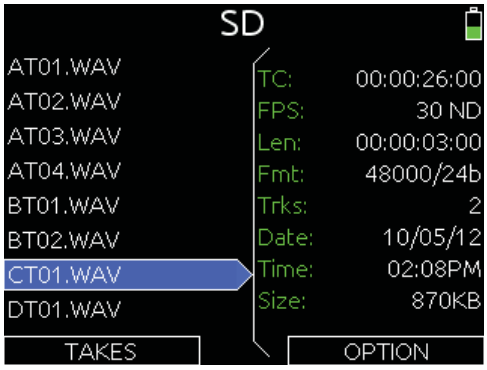
The File List displays all files on the CF or SD card. To access the File List, select Setup Menu option **FILE STORAGE > File List**. The File List can also be accessed by sliding the Slate/Tone Switch while viewing the Take List. When entering the File List, the first view is the Root Menu:

File List: Root Menu

In the Root Menu of the File List, use the Headphone Encoder to highlight either the CF or SD card. Slide the RTN Switch to access options for the highlighted card. Push the Headphone Encoder to display the listing of files and directories on the highlighted card.

When audio files are highlighted, various metadata information is displayed for that file on the right half of the display.

File List: Files on SD card



Slide the RTN Switch to access File Options for the highlighted file, directory, or storage media. The following table lists the File Options that are available for files, directories, and storage media:

File Option	File	Directory	Storage Media
Copy	Copies file to the other storage media.	Copies directory and its contents to the other storage media.	Copies all files and directories to the other storage media.
Delete	Deletes file.	Deletes directory and its contents.	
Create Sound Report		Creates a CSV sound report of all files in the directory and saves the report there. *see Sound Reports*	Creates a CSV sound report of all files in the root of the storage media and saves the report there. *see Sound Reports*
Rename		Changes the name of the directory.	Changes the volume label of the storage media.
Empty Trash		*Available only on TRASH and FALSETAKES directories.* Deletes the contents of the folder.	Deletes the contents of the TRASH directory on the storage media.
Erase/Reformat			Formats the storage media as a FAT32 volume.

File Type

The 664 records audio to the industry-standard Broadcast Wave file format. Additionally, the 664 can record to MP3 format. Recording WAV or MP3 is selectable per media, from the Setup Menu items **RECORDER > Record to CF**, and **RECORDER > Record to SD**. The 664 can write files to both cards simultaneously.

WAV (Broadcast WAV)

The 664 writes AES-31 Broadcast Wave formatted files. The audio files created by the 664 include additional information (Metadata) in the file's header, BEXT (Broadcast Audio Extension) and iXML data chunks. *See Metadata section of Specifications*. Software that does not recognize these additional broadcast wave data chunks will simply ignore them.

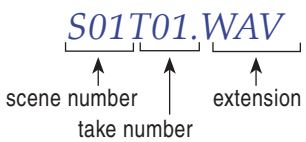
Wave Agent Beta allows for the splitting (de-interleaving) of polyphonic files into separate files for each track.

MP3

MPEG-1 Layer III is a lossy compression algorithm, often used for music and transcription recording purposes. The 664 records two-channel MP3 audio files with data rates of 320 kb/s, 192 kb/s, or 128 kb/s.

File Structure

The 664 records polyphonic audio files. A polyphonic audio file is one file with multiple tracks. When recording polyphonic files, one file will be created per take. A polyphonic take and file name consists of a scene name, take number, and a file extension. For example, for scene S01, take T01, the polyphonic file name and take name are both *S01T01.wav*.



Folder Options

All files generated by the 664 can be saved to the Root directory, a Project Folder, a Daily (Roll) Folder, or a Scene Folder (Files are saved to the Root directory by default). File organization is managed in the Setup Menu option **FILE STORAGE > FOLDER OPTIONS**. Folder choices include **TOP-LEVEL**, **MID-LEVEL**, and **BOTTOM LEVEL FOLDERS**. Folders are created when the record key is pressed. This reduces the likelihood of empty folders.

Each folder resides in the folder one level below it. If the folder one level below the target folder is set to **<None>**, the target folder will reside in the next level folder that is set to something other than **<None>**.

Each folder level has a corresponding list of available names that can be edited manually by the user. To add an entry to the list of folder names of any folder level:

1. Access Setup Menu option **FILE STORAGE > FOLDER OPTIONS**
2. Select desired folder level and press the Headphone Encoder.
3. Rotate the headphone Encoder to select **<Add New Entry>** then press the Headphone Encoder. Use the on-screen keyboard to compose a name for the folder.
4. Slide the RTN Switch right to create the folder name and select it for this folder level.

Top-Level (Project) folder. The Top-Level folder can be named manually or set to **<None>**. The name of the Top-Level folder will also be stored in the PROJECT field of the iXML and bEXT chunks of all subsequently recorded WAV files.

Mid-Level (Tape/Roll/Daily) folder. The Mid-Level folder can be named manually, by the date, or set to **<None>**. When set to **<Daily>**, a folder will be automatically created and named according to the date. Whenever a new day occurs, the 664 will prompt asking if the user would like to create a new daily folder. With an example date of August 20th, 2012, the naming format of the daily folder would be: 12Y08M20. The name of the Mid-Level folder will also be stored in the TAPE field of the iXML and bEXT chunks of all subsequently recorded WAV files.

Bottom-Level folder. The Bottom-Level folder can be named manually, by scene name, or set to **<None>**. When set to **<Scene>**, a new folder will be created each time the scene name is changed.

Automatic File Splitting

A file on a FAT32 volume cannot exceed 4 GB. During recording, the 664 will split a take that exceeds 4 GB into multiple files. The split is sample-accurate, so files joined together in post-production will be seamless. The maximum file size that the 664 creates can be adjusted from the Setup Menu option **FILE STORAGE > File Max Size**. The following file size options are available:

- 4GB
- 2GB
- 1GB
- 640MB
- 512MB

Metadata Fields

664 Broadcast wave files include iXML (revision 1.5) data and bEXT chunk data. For bEXT and iXML-aware software applications this data is available. For applications that don't recognize bEXT or iXML, this information is ignored.

Metadata is included in MP3 files inside the ID3 tags. Metadata in MP3 files can not be edited with the 664.

Sound Reports

The 664 can generate sound reports as a comma separated values (CSV) file. CSV files can be opened and edited by any common spreadsheet application such as Microsoft Excel, OpenOffice Calc, Apple Numbers, Google Docs, and many more.

MP3 files are not included in Sound reports.

Sound Report Setup

A sound report consists of two sections: A Take List section which provides information for each take in the folder the sound report was generated in, and an Overview section that contains ancillary information to the Take List section. The Overview section is first, followed by the Take List section.

Overview Section

The table below lists the rows that make up the Overview section.

Row	Description	Default
Project:	The name of the project.	<NONE>
Producer:	The name of the producer.	<NONE>
Director:	The name of the director.	<NONE>
Job:	The title of the job.	<NONE>
Date:	The date. The <System Date> option will insert the current system date (In MM/DD/YY format) at the time the report is generated.	<System Date>
Location:	The location.	<NONE>
Sound Mixer:	Your name.	<NONE>
Phone:	Your phone number. Limited to numbers and symbols.	<NONE>
Email:	Your email address.	<NONE>
Client:	Client's name.	<NONE>
Boom Op:	Boom operator's name.	<NONE>
Prod.Co:	The name of the production company.	<NONE>
Prod.Co. Tel:	Production company telephone number. Limited to numbers and symbols.	<NONE>
Mics:	Notes about mics used.	<NONE>
Comments:	Miscellaneous comments.	<NONE>
Roll:	The roll / daily identifier.	<NONE>
Media:	Media type	<NONE>
File Type:	File type. Options are <Current Selection> (WAV Poly or MP3) or <None>	<Current Selection>
File Type:	File type. Options are <Current Selection> (WAV Poly or MP3) or <None>	<Current Selection>
Sample Rate:	Sample rate of recorded audio. Options are <Current Selection> (At time of report generation) or <None>	<Current Selection>
Frame Rate:	Timecode frame rate. Options are <Current Selection> (At time of report generation) or <None>	<Current Selection>
Bit Depth:	Bit depth of recorded audio. Options are <Current Selection> (At time of report generation) or <None>	<Current Selection>
Tone Level:	The amplitude of the 664's tone generator. Options are <Current Selection> (At time of report generation) or <None>	<Current Selection>

Information in the Overview section is managed in the Setup Menu option **FILE STORAGE > Sound Report Info**. The 664 derives information for the overview section from these settings at the time the sound report is generated.

Take List Section

The following fields make up the Take List section. Each row of this section corresponds to a take, and each field corresponds directly to the metadata of that take (BEXT and iXML).

- File Name
- Scene
- Take
- Length
- Start TC
- Trk 1 to Trk 16 (16 fields)
- Notes

Generating Sound Reports

A sound report can be generated for all the files in any folder on the 664. Sound reports are created from the File List.

To generate a sound report:

1. Press and hold the Menu Button and then press the Headphone Encoder to access the Take List.
2. Slide the Slate/Tone Switch to access the File List.
3. To create a sound report for the files at the root level of the CF or SD card, highlight the desired card and skip to the next step. To create a sound report for one folder only, use the Headphone Encoder to navigate to the folder and highlight it (do not go into the foldeR).
4. Slide the RTN Switch to access folder options for the highlighted folder.
5. Select the Create Sound Report option. A CSV file will be created in the selected folder. Sound reports created for at the root level of the CF or SD card will be saved as *REPORT.CSV* in the root of that media. Sound Reports created in a subfolder will be named with the last six characters of that folder followed by "*_REPORT.CSV*".

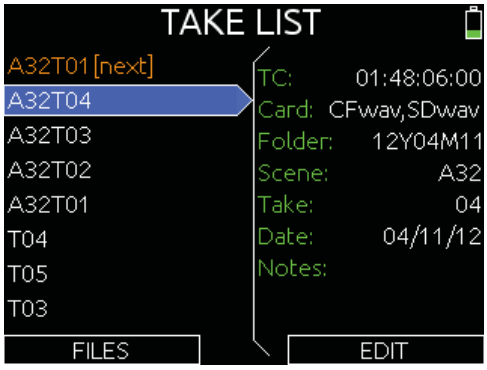
Take Management

Take List

The Take List consists of recorded takes on all media. Takes are listed in the left column in the order they were recorded, with the most recent take at the top. The right column displays details about the highlighted take, including:

- TC: Start timecode value of the take.
- Card: What media the take resides on.
- Folder: The folder that the take resides in.
- Scene: The scene name of the take.
- Take: The take number of the take.
- Date: The date that the take was recorded.
- Notes: The notes metadata field.

To access the Take List press and hold the Menu Button, then press the Headphone Encoder. The Take List can also be accessed from the Setup Menu option **FILE STORAGE > Take List**. The top take in the Take List is displayed with orange text. This is a special entry that represents the next take to be recorded. Editing the metadata of this entry will affect all subsequent recordings.



Take Number

A take is a single recording that may represented by a file on each media. Takes are labeled numerically and begin at the number 1. Each new recording creates a new take. The take number will increment by one until it is reset. Take numbers will be two digits to 99 (01-99) and digits will be added accordingly from 100 and up. By default, take numbers will reset when a new Scene is selected or when a new Daily folder is created. This behavior can be changed with the Setup Menu option **FILE STORAGE > Take Reset Mode**. The take number is written in the file name (preceded by the letter "T") and stored in the metadata of each recorded file. *see File Structure.* The take number

To manually set the take number:

1. Enter the Take List (Menu button + Headphone Encoder push or **FILE STORAGE > Take List**)
2. Highlight the next take (indicated by orange text and the phrase "[next]") and slide the RTN Switch.
3. Rotate the Headphone Encoder to select Take.
4. Using the Headphone Encoder in the numeric entry dialog, enter a new take number. The next take will use this take numbers and subsequent takes will increment from this number until the take number is reset.

To change set the take number of an already recorded take:

1. Enter the Take List (Menu button + Headphone Encoder push or **FILE STORAGE > Take List**)
2. Highlight the take to be edited and slide the RTN Switch.
3. Rotate the Headphone Encoder to select Take. The take number will be changed in the metdata of all files for this take, and all files for this take will be renamed to reflect the new take number.

False Take

To mark a take as a false take and decrement the Take number, press and hold the Headphone Encoder, then slide the Transport Control left (Rewind). A dialog will appear to confirm the false take action. When a take is confirmed as a false take, it will be moved to the FALSETAKES folder on each media and the take number will be decremented by 1. Files will remain in the FALSETAKES folder until they are deleted or the card is formatted.

Scene Name

The scene name is a logical group of takes. It can be any combination of letters or numbers up to 9 characters in length. The scene name will only change when a new scene name is entered by the user or when the scene is incremented. The scene name is written in the file name and stored in the metadata of each recorded file. *see File Structure*

To enter and select a new scene name:

1. Enter the Take List (Menu button + Headphone Encoder push or **FILE STORAGE > Take List**)
2. Highlight the next take (indicated by orange text and the phrase "[next]") and slide the RTN Switch to access the Scene Name List.
3. Rotate the Headphone Encoder to select scene name from the Scene Name List.
4. Using the Headphone Encoder, select an existing scene name and press the Headphone Encoder to select this scene name. To enter a new scene name, select **<Add New Entry>** and enter a new scene name.

To edit the scene name of an already recorded take:

1. Enter the Take List (Menu button + Headphone Encoder push or **FILE STORAGE > Take List**)
2. Highlight the take to be edited and slide the RTN Switch to access the Scene Name List.
3. Rotate the Headphone Encoder to select scene name from the Scene Name List.
4. Enter a new scene name. The scene name will be changed in the metdata of all files for this take, and all files for this take will be renamed to reflect the new scene name.

A scene name can be removed from the Scene Name List by sliding the Slate/Tone Switch right when the scene name is highlighted in the Scene Name List.

Scene Increment

By default Scene names will not increment. When Setup Menu option **FILE STORAGE > Scene Increment Mode** is set to either **Character** or **Numeric**, scene names can be incremented manually by pressing and holding the Headphone Encoder, then sliding the Transport Control right (Fast Forward). If the current scene name does not end in a number or letter, a "1" or "A" will be added (for Numeric or Character increment mode, respectively). If a scene name does end in a letter or number, the letter or number will be incremented.

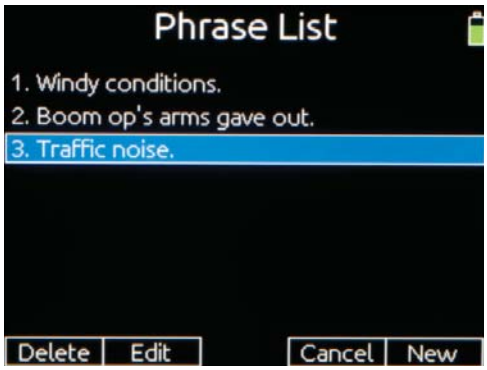
Take Notes and Phrases

The Notes metadata field is used to provide extra information about a take that is not accounted for with the other metadata fields.

Phrases

Often, a particular phrase will be used in the notes field of many different takes. To save data entry time, the 664 provides the ability to save phrases that can be entered into the notes field quickly. To edit, create, and use phrases:

1. Enter the Take List (Menu button + Headphone Encoder push or **FILE STORAGE > Take List**).
2. Highlight the take to be edited, and slide the RTN Switch.
3. Select the Notes field and press the Headphone Encoder.
4. Select **<Use Phrase>** and press the Headphone Encoder to enter the Phrase List.
5. To use a phrase, select it in the list and press the Headphone Encoder. To create a new phrase, slide the RTN Switch right. To edit a phrase, select it in the list and slide the Slate/Tone Switch right. To delete a phrase, select it in the list and slide the Slate/Tone Switch left.



Playback

The 664 can play back any file it has recorded. To play back the last file recorded or loaded, push the Play Button. In the Take List or File List, pressing the Play button will initiate playback of the highlighted take.

When playback begins, the file name will be displayed on the Main Screen with a green background. The timecode and absolute time areas of the Main Screen will display the timecode and absolute time of the file being played back. Pushing the Stop button will pause playback, and pushing the Stop button when playback is paused will stop playback.

The media that files will be played back from is highlighted with a green background on the Main Screen and RTN Screen. To toggle the default playback card, press and hold the Headphone Encoder then press the slide the Transport Control down (Play).

By default, playback will be routed to all outputs of the 664. To limit playback to headphone outputs only, set Setup Menu option **OUTPUTS > Playback to Outputs** to **Headphones Only**.

WAV files recorded on the 664 will play back to tracks as they were recorded. MP3 files recorded on the 664 will always play back to the L and R tracks.

Using a USB Keyboard

To aid in navigation and data entry, a standard USB keyboard can be connected to the 664. To use a USB keyboard, connect the male USB A plug of the keyboard into the supplied USB A female adapter, then connect the keyboard to the left panel USB B jack on the 664 with the supplied USB cable. Change the Setup Menu option **SYSTEM > USB Port** to **Keyboard**.

Keyboards with an embedded USB hub are not compatible. Some keyboards must be disconnected when powering the 664 up and only connected after the 664 has booted.

Whenever the on-screen keyboard is displayed, an attached USB keyboard can be used to enter data. The table below shows the keyboard shortcuts available.

Key Combination	Action
F1	Accesses the Setup Menu.
F2	Accesses the Take List.
F3	Switches between Main Screen and RTN Screen.
Menu Button	Accesses the Setup Menu.
Ctrl + R	Record
Ctrl + S	Stop
Spacebar	Play
Left Arrow	Main Screen: Rewind. Input Settings Screen: Toggle phase inversion.
Right Arrow	Main Screen: Fast Forward. Input Settings Screen: Toggle LR Mix assignment.
Up / Down Arrows	Main Screen: Adjusts headphone volume. In menus: Moves highlighter. While editing parameters: Changes value.

Key Combination	Action
Enter	Main Screen: Accesses HP Monitor Source List. In menus: Same as encoder push (activates current selection).
Alt + Enter	Main Screen: Toggles arming of selected track.
Ctrl + Enter	Accesses gain adjustment for highlighted L, R, X1, X2, or RTN track.
Ctrl + Up / Down Arrows	Main Screen: Select tracks. Input Settings Screen:
Ctrl + Left Arrow	Input Settings Screen: Change X1 assignment.
Ctrl + Right Arrow	Input Settings Screen: Change X2 assignment.

Mixer Linking

Linking the 664 With Another 664 or 552 Mixer

The Link I/O function allows the 664 or 552 to be interconnected with another 664 or 552 mixer for applications requiring more inputs. When the 664 is linked with a 664 or 552, all inputs appear at both mixer's outputs. The stereo mix from the other mixer will be routed to the 664's stereo Master Bus and recorded on tracks L and R. Return audio and PFL occur on the headphone output of both mixers. The Link I/O connector is an unbalanced TA5M wired pin-1 = HP L, pin-2 = HP R, pin-3 = Mix R, pin-4 = Mix L, pin-5 = link detection/ground.

To Link the 664 to another 664 or 552 mixer:

1. Connect the Link I/O of the 664 to the Link I/O of the other 664 or 552 mixer using a TA5F to TA5F cable (Sound Devices XL-TA55 optional accessory). This will send the stereo mix of each mixer directly to the stereo output bus of both mixers.

2. Monitor the stereo sum of all input channels from either mixer headphone monitor.
3. Adjust the levels on each mixer accordingly.

Linking to Other Mixers

The Link I/O function allows the 664 to be linked to a Sound Devices 302, 442, or MixPre mixer for applications requiring higher input channel counts. When the 664 is linked with another Sound Devices mixer, all input channels will be present on the stereo Master Bus of the 664. Return audio and PFL only occur on the headphone output of the mixer that it is performed on. The 552 Link I/O connector is an unbalanced TA5M wired pin-1 = HP L, pin-2 = HP R, pin-3 = Mix R, pin-4 = Mix L, pin-5 = Link detection/ground.

Linking to a 302 or 442

The Tape Out/Mix Out of a 302 is an unbalanced stereo TA3F connector. The 442 has two Tape Out/Mix Out connectors, 3.5 mm and TA3. For the most secure connection use the TA3 Tape Out/Mix Out connector. The TA3 on the 442 and 302 is wired pin-1 = ground, pin-2 = Left, pin-3 = Right. *For details on the 3.5 mm connection see Linking to a MixPre.*

Sound Devices XL-TA35 is a 12-inch, TA5F to TA3F accessory cable used to link Sound Devices 302 and 442 mixers to the 664 or 552. *For more details see Accessories.*

To link a Sound Devices 302 or 442 to a 664:

1. Connect the Link I/O of the 664 to the Tape Out/Mix Out of a 302 or 442 using a Sound Devices XL-TA35 cable. This sends the stereo mix of both mixers directly to the stereo Master Bus of the 664.

2. Monitor and record the stereo sum of all input channels on the Master Bus of the 664.
3. Adjust the levels on each mixer accordingly.

Linking to a MixPre-D or Original MixPre

The MixPre-D's Tape Out 3.5 mm TRS is wired Tip = Left, Ring = Right, Sleeve = ground. Sound Devices XL-TA25 is a 20-inch, TA5F to 3.5 mm TRS accessory cable used to link Sound Devices MixPre-D to the 664. *For more details see Accessories.*

To link a Sound Devices MixPre-D to a 664.

1. Connect the Link I/O of the 664 to the Tape Out of a MixPre-D using a Sound Devices XL-TA25. This sends the stereo mix of both mixers directly to the stereo Master Bus of the 664.

2. Monitor and record the stereo sum of all input channels on the Master Bus of the 664.
3. Adjust the levels on each mixer accordingly.

Powering

The 664 is powered by external DC power or internal AA batteries (NiMH or Lithium chemistries preferred. *see Internal Battery Powering*). The power switch selects the power source, either internal or external. There is a large enough power reserve to switch back and forth from one source to the other without interrupting mixer operation.

External Powering

The 664 can be powered from any DC voltage from 10 to 18 VDC. Pin-4 of the locking, Hirose connector is positive (+) and pin-1 is negative (–). The external DC supply is fully isolated (floating) from the rest of the mixer for easy and safe interconnection with other external audio gear.

Internal Battery Powering

Internal batteries can be used as a back-up in the event that external power is removed or depleted. The 664 uses five AA-sized (LR6) batteries. The 664 can use NiMH, 1.5V Lithium, or Alkaline AA batteries. **Run time will be very limited when using Alkaline batteries (30 minutes to 1 hour, depending on settings). NiMH or Lithium batteries are the preferred type because of the approximate 2x to 4x run time compared to Alkaline batteries.**

Voltage Metering

The Main Screen displays a DC Voltage Indicator in the form of a battery icon that indicates the level of the power source currently in use (internal or external). When the battery icon is filled green, the voltage of the active power source (internal or external) is full. The green color will move down as voltage depletes, changing to red and flashing when the Warning Voltage Threshold is reached. When the voltage is critically low, the entire battery icon will become red and flash.

Maximum Voltage

Minimum Voltage

A numeric reading of the internal and external voltage levels is displayed in the Setup Menu main level at all times. Internal voltage is displayed on the left side and external voltage is displayed on the right side (This example shows a 664 powered from a 12.0V external power supply with no AA batteries inserted):

The voltage range of the DC Voltage Indicator and the Warning Voltage Threshold can be adjusted. These values are adjusted with the Setup Menu option **SYSTEM > Ext DC Ref**. Adjust the Ext DC Reference to match the type of battery being used to power the 664 externally. The voltage range and Warning Voltage Threshold for each setting are shown in the table below.

Setting	Low Voltage	Warning Voltage	High Voltage
12V Ext DC	9	10	11
NiMH	11	11.5	13
Expanded NiMH	11	11.5	18
12V Lead Acid	10	11.4	14
14V Li-ion	12.5	13.5	16.3
Full Range	6	11.5	18

When the voltage of the external power supply drops below the Low Voltage Threshold, the 664 will react differently depending on the state of the internal recorder:

- During recording or playback, the 664 will automatically switch to internal batteries, if available.
- If the internal recorder is idle (Not playing back or recording), the 664 will power down.

During media operations (read or write) the 664 will utilize the main (external) power source if available. This will be reflected on the DC Voltage Indicator.

Power Consumption

The 664 varies in the amount of current it draws. Several features of the 664 directly affect current draw in different ways. The following list highlights the larger current drawing functions.

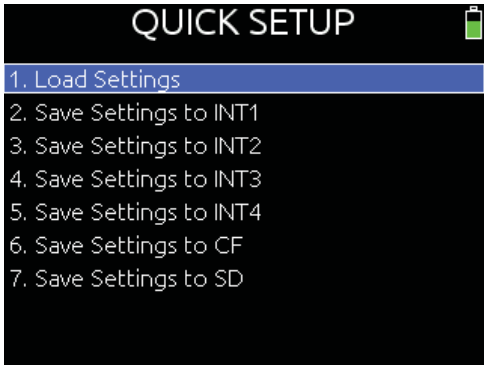
1. Microphone powering - the main source of extra 664 current draw. *See Phantom Power*. 48 V Phantom can draw copious amounts of current depending on what model microphone is used. Two identical phantom powered microphones draw twice as much current as one. Microphones vary widely in their current draw depending on type and phantom voltage applied.

2. Audio Recorder - the recorder, whether in record or playback, increases power consumption.
3. Digital Outputs - disable digital outputs in the Setup Menu when they are not needed since they draw additional current.
4. Output drive level - higher output drive levels into multiple, low-impedance inputs increases power consumption.
5. Headphone output circuit - high headphone output levels increase current draw.
6. LED and LCD Brightness - decrease LED and LCD brightness to extend battery runtime.

Storing and Recalling Settings

The 664 can store and recall user settings to User Setup Files on the SD card, CF card, or internal memory. User Setup Files contain all settings made in the Setup Menu and all Input Settings (including routing). Multiple, custom-named setup files can be saved to the CF or SD card. Additionally, four setups can be saved to the 664's internal memory where they will be available even when no CF or SD card is inserted.

To save a User Setup File, access the Setup Menu option **Quick Setup**. Select the Save option for the desired storage area and enter a name for the Setup File.


QUICK SETUP
1. Load Settings
2. Save Settings to INT1
3. Save Settings to INT2
4. Save Settings to INT3
5. Save Settings to INT4
6. Save Settings to CF
7. Save Settings to SD

- The **INT1**, **INT2**, **INT3**, or **INT4** areas will hold one setup each. Settings stored to these areas will overwrite any settings previously stored there.
- Saving a Setup File to the CF or SD card will create a folder named **SETTINGS** (If it is not already present) and place the Setup File in that directory. If a Setup File with the entered named already exists, the previous Setup File will be overwritten

Setup Files stored on the CF or SD card can be deleted from the File List. To delete a Setup File from the CF or SD card:

1. Press the Menu Button to enter the Setup Menu.
2. Use the Headphone Encoder to select option **FILE STORAGE > File List**.
3. Use the Headphone Encoder to select either CF or SD.
4. Use the Headphone Encoder to select the **SETTINGS** folder.
5. Use the Headphone Encoder to highlight the Setup File to be deleted.
6. Slide the RTN Switch to access File Options for the Setup File.
7. Use the Headphone Encoder to select **Delete**.

Setup Menu

The chart below shows the Setup Menu number, Function name, a description of the control, options available within that menu, and the factory default setting.

Inputs

#	Parameter Name	Description	Options
1	**Ch 1-2 Linking**	Links channels 1 and 2 with or without Mid/Side decoding.	» **Unlinked** • 1-2 • 1-2MS
2	**Ch 3-4 Linking**	Links channels 3 and 4 with or without Mid/Side decoding.	» **Unlinked** • 3-4 • 3-4MS
3	**Ch 5-6 Linking**	Links channels 5 and 6 with or without Mid/Side decoding.	» **Unlinked** • 5-6 • 5-6MS
4	**Phantom Voltage**	Globally adjust voltage level of phantom power (on all inputs which have phantom power enabled).	» **48V** • 12V

Outputs

#	Parameter Name	Description	Options
1	**Linking**	For the purposes of arming and level adjustment, link tracks L and R, and/or X1 and X2.	• LR, X1X2 » **LR, X1, X2** • L, R, X1X2 • L, R, X1, X2
2	**XLR-L Out**	Select the analog output level or engage AES3 digital on the L XLR output.	• Mic » **Line** • -10 • AES 1,2
3	**XLR-R Out**	Select the analog output level or engage AES3 digital on the R XLR output.	• Mic » **Line** • -10 • AES 3,4
4	**10-pin A Out**	Select the analog output level or engage AES3 digital on the both balanced outputs of the 10-pin A output.	• Mic » **Line** • -10 • AES 5-8
5	**10-pin C Out**	Select the analog output level on the both balanced outputs of the 10-pin C output.	• Mic » **Line** • -10
6	**TA3 L-Out**	Select the analog output level of the L TA3 output.	• Mic » **Line** • -10
7	**TA3 R-Out**	Select the analog output level of the R TA3 output.	• Mic » **Line** • -10
8	**TA3-X1 Out**	Select the analog output level of the Aux 1 TA3 output.	• Mic » **Line** • -10

#	Parameter Name	Description	Options
9	**TA3-X2 Out**	Select the analog output level of the Aux 2 TA3 output.	• Mic » **Line** • -10
10	**Direct Out Levels**	Select the analog output level of the TA3 Direct Outputs.	• Mic » **Line** • -10
11	**Direct Out Pre/Post**	Globally select pre- or post-fader signal source for all Direct Outputs.	» **PreFade** • PostFade
12	**Tape Out Source**	Choose the stereo source for the TA3 and 3.5 mm Tape Output.	» **L/R** • RTN A • RTN B • RTN C
13	**AES Output Routing**	Displays the AES output routing matrix where sources can be assigned to AES output channels.	
14	**Playback to Outputs**	Select destination for playback signal.	• Headphones Only » **All Outputs**

Limiters

#	Parameter Name	Description	Options
1	**Limiters**	Globally disable input and output limiters.	• Off » **On**
2	**L,R Limiter Linking**	Links limiters for L and R tracks.	» **On** • Off
3	**L,R Limiter Thresh.**	Limiter threshold for Master Bus	• +4 to +20 dBu (1 dB increment) » **18 dBu**
4	**X1,X2 Limiter Thresh.**	Limtiter threshold for Aux Bus	• +4 to +20 dBu (1 dB increment) » **18 dBu**

Recorder

#	Parameter Name	Description	Options
1	**Record to CF**	Sets the type of file and which tracks to record to the CF card.	• Off » **Wav Poly** • Wav Poly (ISOs Only) • Wav Poly (LR Only) • Wav Poly (X1X2 only) • MP3 (LR) • MP3 (X1X2)
2	**Record to SD**	Sets the type of file and which tracks to record to the SD card.	• Off » **Wav Poly** • Wav Poly (ISOs Only) • Wav Poly (LR Only) • Wav Poly (X1X2 only) • MP3 (LR) • MP3 (X1X2)
3	**MP3 Bit Rate**	The bitrate of recorded MP3 files.	» **320kbs** • 192kbs • 128kbs

#	Parameter Name	Description	Options
4	**Sample Freq.**	The internal sample rate and sample rate of recorded WAV files.	• 44.1k • 47.952k • 47.952kF » **48k** • 48.048k • 48.048kF
5	**Bit Depth**	The bit depth of recorded WAV files.	» **24** • 16
6	**Pre-Roll Time**	Adjust the amount of record time to be appended before each recording.	• 0 to 6 seconds (1 s increment) » **0 s**

Comms/Returns

#	Parameter Name	Description	Options
1	**Slate/Com Mic Source**	Select internal or external slate microphone. Use Ext 12V setting for external condenser microphones.	• OFF » **Int Mic** • Ext Mic • Ext 12V Mic
2	**Ext Slate/Com Mic Gain**	Sets the input gain for external slate microphone.	• 0-36 dB (1 dB increment) » **20 dB**
3	**Com Send Program**	Displays the Com Send Program selection screen where sources can be chosen for right and left channel of the Com Send.	
4	**Com Send Gain**	Sets the output gain of the COM Send output.	• 0-48 dB (1 dB increment) » **0 dB**
5	**Com Return Gain**	Sets the input gain of the COM RTN input.	• 0-24 dB (1 dB increment) » **0 dB**
6	**Program Auto-Mute**	Toggle automatic muting of COM Program when COM communication is activated.	» **Enabled** • Disabled
7	**Mic Toggle Switch**	Selects primary and secondary function of the Slate/Tone Switch.	» **Slate/Com** • Com/Slate
8	**RTN Toggle Left**	Selects primary and secondary function of the left side of the RTN Switch.	» **RTN A/COM RTN** • COM RTN/RTN A
9	**RTN Toggle Right**	Selects primary and secondary function of the right side of the RTN Switch.	» **RTN B/C** • RTN C/B

Timecode/Sync

#	Parameter Name	Description	Options
1	**Sync Reference**	Select synchronization source for digital audio.	» **Internal** • Word Clock
2	**Timecode Mode**	Sets the running mode for the internal timecode generator and/or the method which the 664 receives SMPTE timecode..	• Off • Rec Run » **Free Run** • 24h Run • Ext-TC • Ext-TC Auto-Rec • Ext-TC/Cont. • Ext-TC Auto-Rec/Cont.

#	Parameter Name	Description	Options
3	**Frame Rate**	Sets the frame rate for the internal timecode generator.	• 23.976 • 24 • 25 • 29.97nd • 29.97df » **30nd** • 30df
4	**Hold Off**	Sets the duration that a valid time code signal has to be recognized by the 664 before generating a new take while in Ext TC-Auto Record and Ext TC/cont-Auto Record modes.	• 0.0-8.0 s (0.1 second increment) » **0.0 seconds**
5	**Jam Menu**	Displays the Timecode Jam Screen, where internal, external, and difference timecode can be displayed and the 664 timecode generator jammed to external time code.	
6	**Set Generator TC**	Displays a dialog where the 664 internal timecode generator can be set to a given value. When highlighter is orange, turn Headphone Encoder to highlight a field and press Headphone Encoder to edit. When highlighter is blue, turn Headphone Encoder to change the value, and press Headphone Encoder to accept the value.	
7	**Set Generator UBits**	Sets the timecode user bits of the 664 timecode generator.	
8	**Display Mode**	Selects the source of the large numerical display on the Main Screen.	• Big A-Time » **Big Timecode**

File Storage

#	Parameter Name	Description	Options
1	**Take List**	Displays the Take List.	
2	**File List**	Displays the File List.	
3	**Folder Options**	Allows the user to organize files in root and sub-folders. For no folders, select <None> on every level.	• Top-Level • Mid-Level • Bottom-Level
4	**Sound Report Info**	Allows the user to enter default values for sound report fields. The default values are indicated in parenthesis.	• Project (<None>) • Producer (<None>) • Director (<None>) • Job (<None>) • Date (<System Date>) • Location (<None>) • Sound Mixer (<None>) • Phone (<None>) • E-Mail (<None>) • Client (<None>) • Boom Op (<None>) • Prod. Co. (<None>) • Prod. Co. Tel. (<None>) • Mics (<None>) • Comments (<None>) • Roll (<None>) • Media (<None>) • File Type (CF) (<Current Selection>) • File Type (SD) (<Current Selection>) • Sample Rate (<Current Selection>) • Frame Rate (<Current Selection>) • Bit Depth (<Current Selection>) • Tone Level (<Current Selection>)

#	Parameter Name	Description	Options
5	**Max File Size**	Selects the file size when the 664 will close, then start a new file. The 664 will not record a file larger than the selected size. The largest file permissible with the 664's FAT32 file system is 4 GB	» **4GB** • 2GB • 1GB • 640MB • 512MB
6	**Scene Increment Mode**	Selects which format the scene increments will use. To quickly increment a scene name press and hold the Headphone Encoder then slide the Transport Control right (Fast Forward)	• Disabled • Character • Numeric
7	**Take Reset Mode**	Defines when take numbers are reset. Reset brings take number to <1>	• Never • Scene Change • Daily Folder Change » **Either Scene or Daily**
8	**File Playback Mode**	Determines what (If any) playback action the 664 will perform upon reaching the end of a file during playback.	» **Disabled** • Play All • Repeat One • Repeat All
9	**Default Playback Card**	The source media that files will be played from when playback is initiated.	» **CF** • SD
10	**Erase/Format CF**	Formats CF card to FAT32 file system.	
11	**Erase/Format SD**	Formats SD card to FAT32 file system.	

System

#	Parameter Name	Description	Options
1	**Tone Level**	The level of the 664's internal tone generator.	• 0-20 dBu (1 dBu increment) » **0 dBu**
2	**Tone Frequency**	The audio frequency of the 664's internal tone generator.	• 100-10000 Hz (10 Hz increment) » **1000 Hz**
3	**Tone Mode**	Continuous tone generates tone a the set tone level to all bus tracks. L Ident mode pulses the amplitude of tone to the Left channel of the Master Bus by -20dB.	» **Continuous** • L Ident
4	**Tone Routing**	Sets the destination of the tone generator.	» **LR+Direct Outs** • LR+X1+Direct Outs • LR+X2+Direct Outs • LR+X1X2+Direct Outs
5	**Warning Bell Level**	The level of warning bell signals in the Headphones.	• Off, -60 to -12 dBFS (1dB increment) » **-30 dBFS**
6	**Record/Stop Bell**	Alerts the user with one beep at the start of recording and two beeps when the recording is stopped	• Off » **On**
7	**Meter Ballistics**	Selects among the two different meter ballistics settings.	• VU Only » **Peak + VU** • Peak Only
8	**Meter Peak Hold Time**	Adjusts the duration for which the peak indicator stays illuminated.	» **1 sec** • 2 sec • 3 sec • 4 sec • 5 sec • Infinity • Off

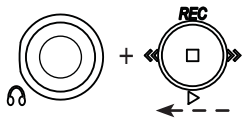
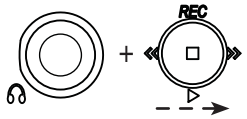
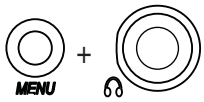
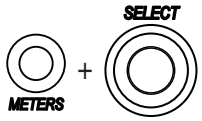
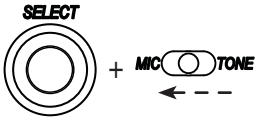
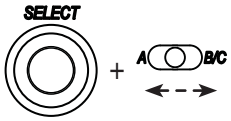
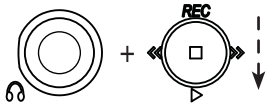
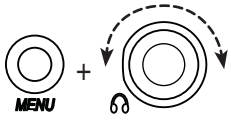
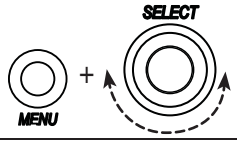
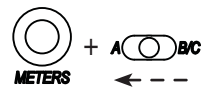
#	Parameter Name	Description	Options
9	**LED Brightness**	Adjust the brightness of all LED's.	• 5-100% (5% increment) » **60%**
10	**LCD Brightness**	Adjusts the brightness of the LCD backlight.	• 5-100% (5% increment) » **100%**
11	**Ext DC Ref**	Adjust the voltage range and Warning Voltage for the external DC Voltage Indicator.	» **12V Ext DC** • NiMH • Expanded NiMH • 12V Lead Acid • 14V Li-ion • Full Range
12	**Time Format**	Select 12 hour clock or 24 hour clock.	» **12hr** • 24hr
13	**Date Format**	Select the format used to indicate the date.	» **mm/dd/yy** • dd/mm/yy • yy/mm/dd
14	**Set Time/Date**	Displays a dialog where the 664 time and date can be set. When highlighter is orange, turn Headphone Encoder to highlight a field and press Headphone Encoder to edit. When highlighter is blue, turn Headphone Encoder to change the value, and press Headphone Encoder to accept the value.	
15	**Fader/Pan Calibration**	Calibrates fader and pan positions. Set all faders and pans to center position and press OK.	
16	**Track Names in Meters**	Displays track names on top of the meters of the Main Screen and RTN Screen	» **Off** • On
17	**USB Port**		» **Keyboard** • Factory Test
18	**Shortcut Info**	Displays a list of all the 664 button shortcut combination for reference.	
19	**Version Info**	Displays firmware version information and Serial Number of the 664.	
20	**Update Firmware**	Searches connected media for .prg files and prompts to begin firmware update for each file found.	

Quick Setup

#	Parameter Name	Description	Options
1	**Load Settings**	Lists all available setup files found on attached media and on internal setup storage.	» **Factory Default** • Other available setups....
2	**Save Settings to INT1**	Saves settings to internal setup storage location 1.	
3	**Save Settings to INT2**	Saves settings to internal setup storage location 2	
4	**Save Settings to INT3**	Saves settings to internal setup storage location 3	
5	**Save Settings to INT4**	Saves settings to internal setup storage location 4	
6	**Save Settings to CF**	Saves settings to CF card.	
7	**Save Settings to SD**	Saves settings to SD card	

Front Panel Button Shortcuts

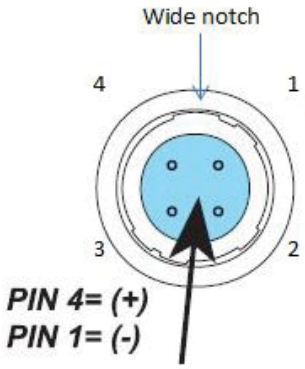
To speed navigation the 664 has numerous navigation "shortcuts". For combinations, press and hold down the first identified key then press the next identified key(s).

Function	Key Sequence	Action
False Take		Press and hold the Headphone Encoder then slide the Transport Control left (Rewind). Moves take to FALSETAKES folder and decrements take number.
Scene Increment		Press and hold the Headphone Encoder then slide the Transport Control right (Fast Forward). Increments the scene name according to the Scene Increment Mode (Setup Menu option **FILE STORAGE > Scene Increment Mode**)
Take List		Press and hold the Menu button then press the Headphone Controller Enters the Setup Menu.
Arm / Disarm Track		Press and hold the Meters button then press the Select Encoder. Toggles the armed status of the highlighted track on the Main Screen.
Alternate Slate function		Press and hold the Select Encoder then slide the Slate/Tone Switch left. Selects the alternate slate function (per Setup Menu option **COMMS/RETURNS > Mic Toggle Switch**).
Alternate RTN		Press and hold the Select Encoder then slide the RTN Switch left or right. Selects the alternate RTN source (per Setup Menu option **COMMS/RETURNS > RTN Toggle Left** and **COMMS/RETURNS > RTN Toggle Right**)
Toggle Playback Card		Press and hold the Headphone Encoder then slide the Transport Control down (Play). Changes the target media between SD or CF. Target media is indicated by a green background on the Main Screen.
LED Brightness		Press and hold the Menu button then rotate the Headphone Encoder. Adjusts the brightness of all LED's.
LCD Brightness		Press and hold the Menu button then rotate the Select Encoder. Adjusts the backlight level of the LCD.
Frame rate / Sample rate display toggle		Press and hold the Meters button then slide the RTN Switch left. Toggles display of Timecode frame rate information and Audio sample rate information between Main Screen and RTN Screen.

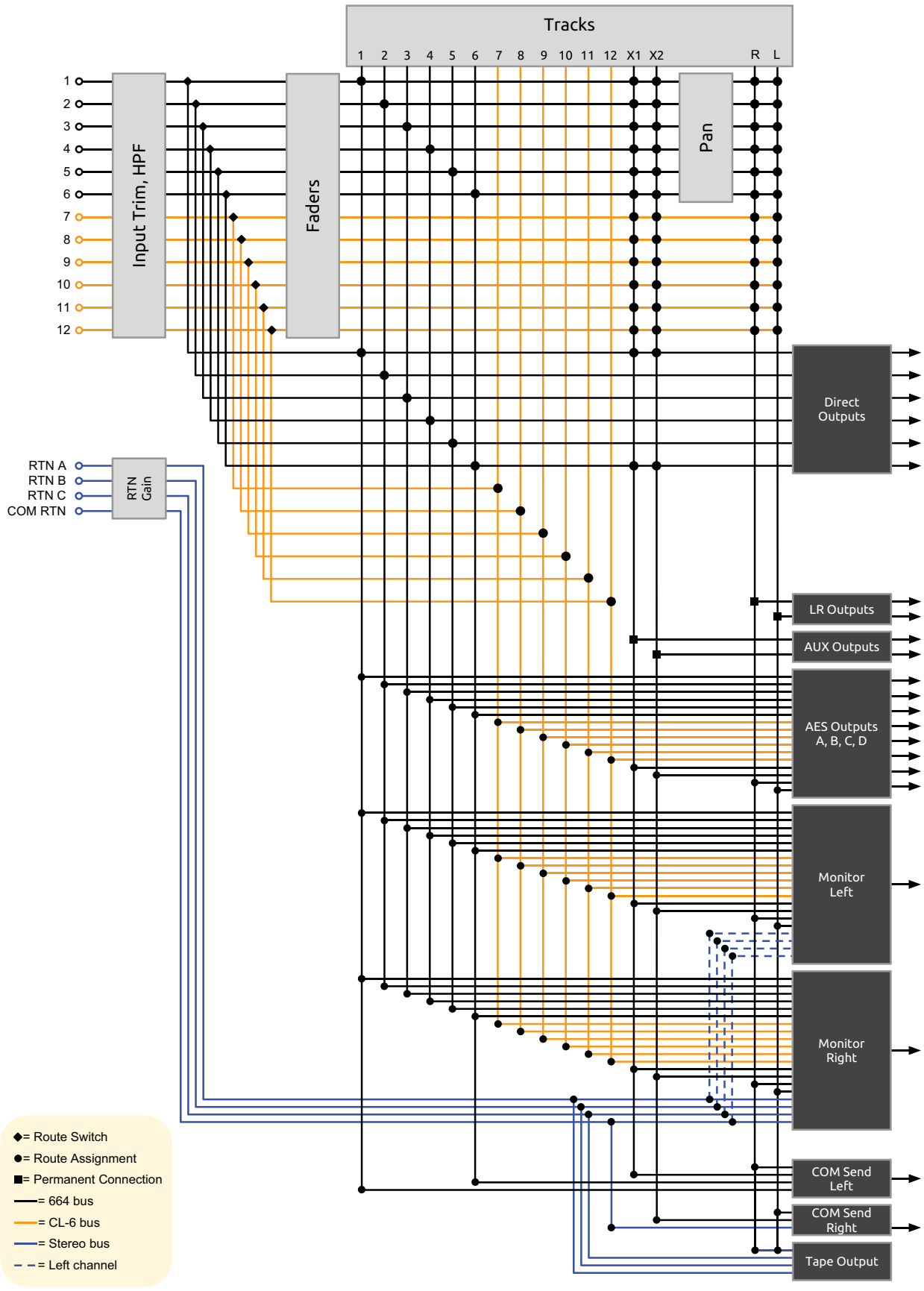
Connector Pin Assignments

Connector		Pin Assignments	Notes
(6) XLR F **Analog Inputs 1-6**		1 – ground 2 – signal (+) 3 – signal (-)	Mic- and Line-level balanced. For unbalanced, pin 1 and pin 3 tied together = ground, pin 2 = positive.
(6) TA3M **Direct Outputs 1-6 / CL-6 Inputs 7-12**		1 – ground 2 – signal (+) 3 – signal (-)	Active balanced direct outputs (without CL-6) or active balanced inputs 7-12 (with CL-6). Output level is selectable in Setup Menu as Line, -10, or Mic level. For unbalanced, pin 1 = ground, pin 2 = positive, pin 3 = float (no connect). Mates with Switchcraft TA3F-type connector.
(1) TA3M **Slate Mic Input**		1 – ground 2 – signal (+) 3 – signal (-)	Active balanced input for external Slate Microphone.
(2) XLR M **Master Outputs/ AES Outputs**		1 – ground 2 – signal (+) 3 – signal (-)	Transformer balanced, Output Level is selectable via menu between Mic, -10 dB, or Line-level. For unbalanced, pin 1 and pin 3 tied together = ground, pin 2 = positive. Balanced AES3 Out A,B on XLR connectors, 110 ohm, 2 V p-p, AES3 and S/PDIF compatible with RCA adaptor.
(1) 3.5 mm **RTN B**		tip – signal L ring – signal R sleeve – signal ground	Mates with 3.5 mm TRS jack. Signal is unbalanced.
(2) 10-Pin Hirose F **Master Output / RTN Input**		1 – L (+) output 2 – L (–) output 3 – R (+) output 4 – R (–) output 5 – R (+) return 6 – n/c 7 – L (+) return 8 – n/c 9 – ground 10 – ground	Mates with Hirose RM15TPD-10P(71) connector. Outputs are Transformer Balanced, output level is menu selectable between Line, -10, or Mic level. For unbalanced connections XLR connectors should be wired, pin 1 and pin 3 tied together = ground, pin 2 = positive. Balanced AES3 Out C,D on top 10-Pin connector, 110 ohm, 2 V p-p, AES3 and S/PDIF compatible with RCA adaptor.
(4) TA3M **L, R and X1, X2 Outputs**		1 – ground 2 – signal (+) 3 – signal (-)	Active balanced, output level is selectable in Setup Menu as Line, -10, or Mic level. For unbalanced, pin 1 = ground, pin 2 = positive, pin 3 = float (no connect). Mates with Switchcraft TA3F-type connector.
(1) TA3M **COM RTN**		1 – ground 2 – signal (+) 3 – signal (-)	Balanced line input for COM RTN signal. Mates with Switchcraft TA3F-type connector
(1) TA3M **COM Send**		1 – ground 2 – Left Signal 3 – Right Signal	Unbalanced stereo output. Mates with Switchcraft TA3F-type connector.
(1) TA3M **Tape Output**		1 – ground 2 – Left Signal 3 – Right Signal	Unbalanced stereo output. Mates with Switchcraft TA3F-type connector.
(1) 3.5 mm **Tape Output**		tip – signal L ring – signal R sleeve – signal ground	Mates with 3.5 mm TRS jack. Signal is unbalanced.
(1) 5-Pin **Mixer Link I/O**		1 – HP L 2 – HP R 3 – Mix R 4 – Mix L 5 – Link detection/Ground	Used to link the 664 to Sound Devices MixPre, MixPre-D, 302, 442, 552, or 664 Field Mixers. Mates with Sound Devices XL-TA25, XL-35, and XL-TA55 optional accessory cables. See Accessories for details. Custom cable note: cable **must** have shield connected to shell on both sides and a 10K resistor from pin 5 to shield. See XL-TA55 documentation for details.

Connector	Pin Assignments	Notes
(1) 3.5 mm Headphone	tip – signal L ring – signal R sleeve – signal ground	Mates with 3.5 mm TRS jack.
(1) 1/4-inch Female Headphone	tip – signal L ring – signal R sleeve – signal ground	Mates with 1/4-inch TRS jack.
(1) 5-pin LEMO Time code	1 – ground 2 – SMPTE TC In 3 – ASCII in/out 4 – tuning out 5 – SMPTE TC out	LEMO B-series connector, pin assignments as viewed on panel-mounted connector
(1) Hirose 4-pin DC Input	1 – ground 2 – not connected 3 – not connected 4 – DC (+)	10-18 Volt DC input. Mates with Sound Devices XL-NPH and XL-WPH3 powering accessories. See Accessories for details. This is a floating supply, isolated from the mixer's ground. See below illustration for pin identification.

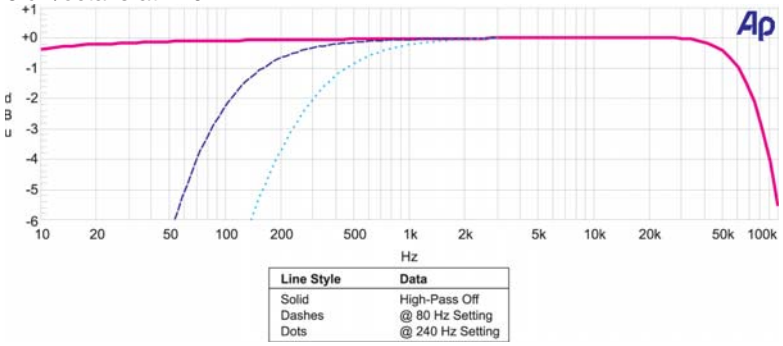


Routing Diagram



Specifications

Analog Inputs

Frequency Response	10 Hz to 50 kHz, ± 0.5 dB
THD + Noise	0.09% max (1 kHz, 22 Hz–22 kHz BW, fader at 0, 0 dBu output)
Equivalent Input Noise	-126 dBu (-128 dBV) maximum. (22 Hz - 22 kHz bandwidth, flat filter, trim control fully up)
Input Type	XLR Mic: Active-balanced for use with ≤600 ohm mics XLR Line: Active-balanced for use with ≤2k ohm outputs RTN A,B,C (3.5 mm/10-pin): Unbalanced stereo for use with ≤2k ohm outputs Link I/O: Unbalanced stereo for linking to Sound Devices MixPre, 302, 442, 552, and 664.
Input Impedance (ohms actual)	XLR Mic: 4k ohms XLR Line: 10k ohms RTN A,B,C (3.5 mm/Multi-pin): 30k ohms Link I/O: 2k ohms
Input Clipping Level	0 dBu minimum (trim control fully down)
Maximum Input Level	XLR Mic: 0 dBu (0.78 Vrms) XLR Line: +40 dBu (80 Vrms) RTN A,B,C (3.5 mm/Multi-pin): +24 dBu (12.4 Vrms) Link I/O: +6 dBu (1.5 Vrms)
High-Pass Filters	Sweepable 80 Hz to 240 Hz, 12 dB/oct at 80 Hz, 6 dB/octave at 240 Hz 
Microphone Powering (each analog Input selectable)	Dynamic (no power applied), 12 V Phantom - through 680 ohm resistors, 10 mA per mic available, 48 V Phantom - through 6.8k resistors, 10 mA per mic available
Input Limiters	Affects the output of the mic preamps only ('Trim' stage), +16 dBu threshold, 20:1 limiting ratio, 1 mS attack time, 500 mS release time.

Analog Outputs

Output Type	XLR Line: Transformer-balanced for use with ≥600 ohm inputs XLR -10: Transformer-balanced for use with ≥10k ohm inputs XLR Mic: Transformer-balanced for use with ≥600 ohm inputs 10-pin Hirose Line: Transformer-balanced for use with ≥600 ohm inputs 10-pin Hirose -10: Transformer-balanced for use with ≥10k ohm inputs 10-pin Hirose Mic: Transformer-balanced for use with ≥600 ohm inputs TA3 Active Balanced Mic/Line: pin-2 and 3 driven, for use with ≥3k ohm inputs TA3 Active Balanced Direct Outs Mic/Line: pin-2 and 3 driven, for use with ≥3k ohm inputs Tape Outs (3.5 mm and TA3-type): Unbalanced, stereo, for use with ≥6k ohm inputs Headphones (3.5 mm and 1/4"): Unbalanced, stereo, for use with 8-2k ohm headphones
Output Impedance (ohms actual)	XLR: 100 ohms at Line setting, 3.2k ohms at -10 setting, 150 ohms at Mic setting 10-pin Hirose: 100 ohms at Line setting, 3.2k ohms at -10 setting, 150 ohms at Mic setting TA3 Balanced Outs: 1k ohms at Mic and Line settings TA3 Direct Outs: 1k ohms at Mic and Line settings Tape Outs (3.5 mm and TA3-type): 1.8k ohms Headphones (3.5 mm and 1/4"): 50 ohms

Line Output Clipping Level (1% THD)	20 dBu minimum with 10k load
Maximum Output Level	Line (XLR, 10-pin Hirose, TA3 Balanced and Direct Outs): +20 dBu (7.8 Vrms) -10 (XLR, 10-pin Hirose): +6 dBu (1.5 V rms) Mic (XLR, 10-pin Hirose, TA3 Balanced and Direct Outs): -20 dBu (0.078 Vrms) Tape Outs (3.5 mm and TA3-type): +6 dBu (1.5 Vrms)
Output Limiters	Affects the outputs of the mixer. Threshold selectable from +4 dBu to +20 dBu, 1 dB steps, 20:1 limiting ratio, 1 mS attack time, 500 mS release time.

Maximum Gain
Typical (Trim, Fader, Master, Phones, RTN A, RTN B, RTN C fully up)

	MIC Level Input	LINE Level Input	RTN A/B/C
Master Outs (XLR, 10-pin, TA3) @ LINE	93 dB	53 dB	-
Master Outs (XLR, 10-pin, TA3) @ -10	79 dB	39 dB	-
Master Outs (XLR, 10-pin, TA3) @ MIC	53 dB	13 dB	-
Direct Outs Post-Fade (TA3) @ LINE	87 dB	47 dB	-
Direct Outs Post-Fader (TA3) @ -10	73 dB	33 dB	
Direct Outs Post-Fader (TA3) @ MIC	47 dB	7 dB	-
Direct Outs Pre-Fader (TA3) @ LINE	72 dB	32 dB	-
Direct Outs Pre-Fader (TA3) @ -10	58 dB	18 dB	-
Direct Outs Pre-Fader (TA3) @ MIC	32 dB	-8 dB	-
Tape Out (TA3 and 3.5 mm)	79 dB	39 dB	6 dB
Headphones (1/4" and 3.5 mm)	107 dB	67 dB	34 dB

Digital Outputs/Recorder

AES3 Output:	Transformer-balanced AES Out A,B on XLR and 10-pin Hirose connectors, AES Out C on Hirose 10-pin connector 110 ohm, 2 V p-p, AES and S/PDIF compatible with RCA adaptor
Sampling Frequency	44.1 kHz, 47.952 kHz, 48 kHz, 48.048 kHz
A/D	24 bit
A/D Dynamic Range	114 dB, A-weighted, typical
Media Type	Secure Digital High Capacity (SDHC), Secure Digital (SD), CompactFlash (CF) FAT32 formatted, will format media on-board
File Type	Record: WAV (Broadcast Wave File format), polyphonic Playback: WAV (Broadcast Wave File format), polyphonic
Sampling Clock Accuracy	± 0.2ppm

Timecode and Sync

Modes Supported	Off, Rec Run, Free Run, 24h Run, External
Frame Rates	23.976, 24, 25, 29.97DF, 29.97ND, 30DF, 30ND
Accuracy	Ambient Generator, 0.2ppm (0.5 frames per 24 hours) accuracy. Holds TC clock for two hours after main battery removal
Input / Output	Input: 20k ohm impedance, 0.3 V - 3.0 V p-p (–17 dBu - +3 dBu) Output: 1k ohm impedance, 3.0V p-p (+12 dBu)
Sync Input	Word Clock
Word Clock Output	Square wave, running at sample rate, 3.3 V p-p, 75 ohm.

Power

External Power Supply	Isolated (floating) DC input jack, 10-18 V Locking 4-pin Hirose connector, pin-4 = (+), pin-1 = (-). Mates with gold Hirose #HR10A-7P-4P (DigiKey# HR110-ND) or silver Hirose #HR10-7P-4P (DigiKey# HR100-ND) locking connector.
Internal Power Supply	5-10 V range. Internal backup batteries. Accepts 5 AA-sized (LR6) batteries (NiMH re-chargeable only). There are similar sized 3+ V Lithium-cells that are not compatible.

Environmental

Operation and Storage	Operating: -20°C to 60°C, 0 to 90% relative humidity; (non-condensing) Storage: -40°C to 85°C

Dimensions and Weight

Size	5.3 cm x 32 cm x 19.8 cm (H x W x D) 2.1” x 12.6” x 7.8”
Weight	4 lbs. 12 oz. (unpackaged, without batteries)

Accessories

Several high-value accessories are available for the 664, including production cases, cables, and power accessories. For a full list of Sound Devices products and accessories, visit our web site **www.sounddevices.com/products**.

Optional 664 Related Accessories

CS-664	Production case with high-quality strap for use with the 664 Field Production Mixer with CL-6 Input Expander; accommodates NP type battery and includes doors to access word clock and talkback send/receive connection. Includes large detachable wireless bag; by CamRade for Sound Devices.
Wave Agent Beta	Sound Devices Wave Agent file librarian for Mac OS and Windows computers. Wave Agent provides a comprehensive and indispensable range of tools for preparing audio files for problem-free passage through complex production workflows. For more details and download visit *www.waveagent.com*.
XL-1B	12-inch TA3F to TA3F cable, connects the 664 Direct or TA3 Aux and Master Bus outputs to TA3 inputs of receiving devices such as the Sound Devices 788T's analog inputs 5-8. XL-1B is also used for 664 unbalanced stereo Tape Out and RTN B Input.
XL-2	25-inch TA3F to XLR-M cable, used to connect 664 Direct or TA3 Aux and Master Bus outputs to third-party devices with XLR-F inputs; package of two.
XL-2F	25-inch XLR-F to TA3F cable, used to connect mixers and other devices with XLR-M outputs to the 664 TA3 7-12 inputs or External Slate mic input; package of two.
XL-4	Bag of four (4) TA3-F-type connectors.
XL-10	Hirose 10-pin to two-XLR (balanced L/R) and 3.5 mm plug (664 Stereo Return A and C) breakout cable, 24-inch; includes in-line 20-foot extension cable; 442, 552, and 664-specific.
XL-H	Bare Hirose 4-pin locking DC connector (HR10-7P-4P).
XL-NPH	NP-type battery cup with 24-inch cable terminated in Hirose 4-pin locking DC connector (HR10-7P-4P) at equipment end.
XL-TA25	TA5F to 3.5 mm TRS, 12-inch, used to connect 664 TA5M Link I/O to the MixPre or 442 Tape Out/Mix Out 3.5 mm jack for mixer linking.
XL-TA35	TA5F to TA3F, 12-inch, used to connect 664 TA5M Link I/O to 302 or 442 Tape Out/Mix Out TA3M connector for mixer linking.
XL-TA55	TA5F to TA5F, 12-inch, used to link the 664 to another 664 or 552 mixer.
XL-WPH3	AC to DC Power Supply (in-line) 100 - 240V 50/60 Hz input, 12 VDC 3.75 A (45 W) output, Hirose 4-pin DC plug. Supplied with 3-pin IEC cord.

Wave Agent

Wave Agent Beta is a WAV file librarian for Mac OS and Windows computers. Designed for Production Sound Mixers and Post Production Editors, Wave Agent provides a comprehensive and indispensable range of tools for preparing audio files for problem-free passage through complex production workflows.

Wave Agent is an essential, time-saving tool for anyone working in Production Sound. It allows for the following;

- Playback of polyphonic and monophonic WAV and Broadcast Wave files from any source
- Viewing and editing file metadata
- Changing of sampling rate, frame rate, and start time code metadata stamps
- Splitting and combining polyphonic and monophonic files with selectable track assignments
- Batch editing
- Generating customized PDF and CSV Sound Reports
- Large time code display
- Monitor mixer with faders, pans, solos, mutes, and meters
- Drag-and-drop operation
- Comprehensive keyboard shortcuts for accelerated operation
- Mac OS and Windows compatible

Wave Agent is available as a no-charge software download for either Mac OS or Windows.
http://www.sounddevices.com/products/waveagent/

SOUND DEVICES	**Declaration of Conformity** According to EN ISO/IEC 17050-1:2004	CE

Manufacturer's Name: Sound Devices, LLC

Manufacturer's Address: E7556 State Rd. 23 and 33
Reedsburg, WI 53959
USA

Declares under sole responsibility that the product as delivered

Product Name: 664 Field Mixer with Integrated Recorder

Model Number: 664

Product Options: This declaration covers all options of the above products

complies with the essential requirements of the following applicable European Directives, and carries the CE marking accordingly:

EMC Directive (2004/108/EC)

EN 55022:2006 + A1:2007

EN 55103-2:2009

First date of CE approval August 29, 2012.

This Declaration of Conformity applies to the above-listed products placed on the EU market after:

August 29, 2012
Date

Matt Anderson
Director of Engineering

Warranty and Technical Support

Warranty & Service

Sound Devices, LLC warrants the 664 against defects in materials and workmanship for a period of ONE (1) year from date of original retail purchase. This is a non-transferable warranty that extends only to the original purchaser. Sound Devices, LLC will repair or replace the product at its discretion at no charge. Warranty claims due to severe service conditions will be addressed on an individual basis. THE WARRANTY AND REMEDIES SET FORTH ABOVE ARE EXCLUSIVE. SOUND DEVICES, LLC DISCLAIMS ALL OTHER WARRANTIES, EXPRESS OR IMPLIED, INCLUDING WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE. SOUND DEVICES, LLC IS NOT RESPONSIBLE FOR SPECIAL, INCIDENTAL, OR CONSEQUENTIAL DAMAGES ARISING FROM ANY BREACH OF WARRANTY OR UNDER ANY OTHER LEGAL THEORY. Because some jurisdictions do not permit the exclusion or limitations set forth above, they may not apply in all cases.

For all service, including warranty repair, please **contact Sound Devices for an RMA** (return merchandise authorization) before sending your unit in for repair. Product returned without an RMA number may experience delays in repair. When sending a unit for repair, please do not include accessories, including SD cards, batteries, power supplies, carry cases, cables, or adapters unless instructed by Sound Devices.

Sound Devices, LLC
Service Repair RMA #XXXXX
E7556 State Rd. 23 and 33
Reedsburg, WI 53959 USA
telephone: (608) 524-0625

Technical Support / Bug Reports

For technical support and bug reporting on all Sound Devices products contact:

Sound Devices, LLC
E-mail: support@sounddevices.com
web: http://www.sounddevices.com/support/
Telephone: +1 (608) 524-0625 / Toll-Free in the U.S.A.: (800) 505-0625
Fax: +1 (608) 524-0655

Sound Devices cannot guarantee that a given computer, software, or operating system configuration can be used satisfactorily with 664 generated files based exclusively on the fact that it meets our minimum system requirements. Please check with your software editing application to make certain that it is compatible with the file type selected

---

SOUND DEVICES